



molekular

Manual



The information in this document is subject to change without notice and does not represent a commitment on the part of Native Instruments GmbH. The software described by this document is subject to a License Agreement and may not be copied to other media. No part of this publication may be copied, reproduced or otherwise transmitted or recorded, for any purpose, without prior written permission by Native Instruments GmbH, hereinafter referred to as Native Instruments.

“Native Instruments”, “NI” and associated logos are (registered) trademarks of Native Instruments GmbH.

Mac, Mac OS, GarageBand, Logic, iTunes and iPod are registered trademarks of Apple Inc., registered in the U.S. and other countries.

Windows, Windows Vista and DirectSound are registered trademarks of Microsoft Corporation in the United States and/or other countries.

All other trade marks are the property of their respective owners and use of them does not imply any affiliation with or endorsement by them.

Document authored by: Gustav Sto. Tomas

Software version: 1.0 (03/2014)

Special thanks to the Beta Test Team, who were invaluable not just in tracking down bugs, but in making this a better product.

Germany

Native Instruments GmbH
Schlesische Str. 29-30
D-10997 Berlin
Germany
www.native-instruments.de

USA

Native Instruments North America, Inc.
6725 Sunset Boulevard
5th Floor
Los Angeles, CA 90028
USA
www.native-instruments.com

Japan

Native Instruments KK
YO Building 3F
Jingumae 6-7-15, Shibuya-ku,
Tokyo 150-0001
Japan
www.native-instruments.co.jp



© Native Instruments GmbH, 2014. All rights reserved.

Table of Contents

1	Welcome to MOLEKULAR	8
1.1	Basic Information	8
1.2	About MOLEKULAR	9
2	Installation and Activation	10
2.1	Installing MOLEKULAR	10
2.2	Activating MOLEKULAR	10
3	How to Use MOLEKULAR	12
3.1	How to Open MOLEKULAR	12
3.2	Exploring Factory-set Snapshots	15
3.2.1	Loading a Snapshot from the Sidepane	16
3.2.2	Loading a Snapshot from the Header	17
3.3	Saving a Snapshot	17
3.4	Selecting MOLEKULAR A and B Panel Views	17
4	Overview of the MOLEKULAR Interface	20
4.1	View B Controls	20
4.2	View A Sections	22
5	Modulation	24
5.1	Assigning Modulation	24
5.2	Assigning Cross Modulation	27
5.3	Managing Modulation via the Modulation Screen	31
5.4	Removing Modulation Destinations	37
5.5	Clearing All Modulation	38
5.6	Stereo Modulation	40
5.7	Modulation Unit Reference	42
5.7.1	LFO	42
5.7.2	Step Sequencer	45

5.7.3	Envelope	47
5.7.4	Logic	50
6	Macro Controls	59
7	Morphing	62
7.1	The Morph Controls	62
7.2	Morph Assignment	63
7.2.1	Selecting Available Destinations	63
7.2.2	Assigning a Morph Base and Offsets	65
7.2.3	Removing Morph Base or Offsets	70
7.2.4	Removing Morph Destinations	72
7.3	Morph Modulation: Motion	73
7.3.1	Managing Motion via the Morph Modulation Screen	77
7.4	Morph Quantization	81
8	Routing, Patching, and Pitch Quantization	83
8.1	The Routing Selector	85
8.2	Using the Routing Section	88
8.2.1	Swapping DSP Positions	89
8.2.2	Adding a Crossfader	89
8.3	Patching	91
8.4	Pitch Quantization	93
8.4.1	Applying Pitch Quantization	96
8.4.2	Modulating Pitch Quantization	101
9	DSP Reference	102
9.1	Using the Effect Units	102
9.1.1	The Header	103
9.1.2	The Footer	104
9.2	Recurring Effect Units	105
9.2.1	Dual Delay	105

9.2.2	Equalizer	106
9.2.3	Filter	107
9.2.4	Level	108
9.2.5	Metaverb	111
9.3	DSP 1 Effect Units	112
9.3.1	Dual Comb	112
9.3.2	Plagiarism	113
9.3.3	reSonitarium	115
9.3.4	Spektral Hold	118
9.3.5	Spektral Shift	120
9.3.6	Spektral Smear	121
9.3.7	Vokoder	123
9.4	DSP 2 Effect Units	124
9.4.1	Angel Delay	124
9.4.2	Band Delays	126
9.4.3	Cloud Delay	128
9.4.4	Dub Delay	130
9.4.5	Freezer	131
9.4.6	Iteratron	133
9.4.7	Reverseoid	135
9.4.8	Ryuchi	136
9.4.9	Trails	137
9.5	DSP 3 Effect Units	139
9.5.1	Chorus	139
9.5.2	Dark Forces	140
9.5.3	Dual Filter	143
9.5.4	Filterbank	144
9.5.5	Frequency Shift	145

9.5.6	Half Wave	146
9.5.7	Phlanger	147
9.5.8	Pitch Shift	149
9.6	DSP 4 Effect Units	150
9.6.1	Lo Fi	150
9.6.2	Modulo Fry	151
9.6.3	Slam Dunk	153
9.6.4	Track OSC	154
9.6.5	Track Pulses	156
9.6.6	Wave Fold	157
10	Credits	159

1 Welcome to MOLEKULAR

1.1 Basic Information

Thank you very much for downloading this REAKTOR ensemble from Native Instruments. This new and exciting effect application can be used either with the free REAKTOR PLAYER, or the full version of REAKTOR 5.9 (or above). On behalf of the entire Native Instruments team, we hope this product will inspire you.



To get the best from this instrument please read the manual in its entirety.

Manual Conventions

This manual uses particular formatting to point out special facts and to warn you of potential issues. The icons introducing the following notes let you see what kind of information is to be expected:



Whenever this exclamation mark icon appears, you should read the corresponding note carefully and follow the instructions and hints given there if applicable.



This light bulb icon indicates that a note contains useful extra information. This information may often help you to solve a task more efficiently, but does not necessarily apply to the setup or operating system you are using; however, it's always worth a look.

Furthermore, the following formatting is used:

- Text appearing in (drop-down) menus (such as *Open...*, *Save as...* etc.) and paths to locations on your hard drive or other storage devices is printed in *italics*.
- Text appearing elsewhere (labels of buttons, controls, text next to checkboxes, etc.) is printed in **light blue**. Whenever you see this formatting applied, you will find the same text appearing somewhere on the screen.
- Important names and concepts are printed in **bold**.
- Single instructions are introduced by this play button type arrow.

- Results of actions are introduced by this smaller arrow.

1.2 About MOLEKULAR

MOLEKULAR will change the way you use effects. Design your own modular system with 35 exclusive effects and set it in motion with powerful modulation. Perform your effects with the interactive morphing field and control the entire system with macros. MOLEKULAR will expand your sonic imagination – get ready to be inspired.

2 Installation and Activation

2.1 Installing MOLEKULAR

The following section explains how to install and activate MOLEKULAR. Although this process is straightforward, please take a minute to read these instructions, as doing so might prevent some common problems.

- ▶ To install MOLEKULAR, double-click the installer application and follow the instructions on the screen. The installer application automatically places the new ensemble file into a REAKTOR PLAYER directory. Alternatively, during the installation process, choose the directory where you would like to have MOLEKULAR installed.

The full version of REAKTOR (5.9 or later) or the free REAKTOR PLAYER is required to play REAKTOR Instruments and Effects. You can download the free REAKTOR PLAYER from the Native Instruments website.

2.2 Activating MOLEKULAR

When installation is finished, start the Service Center application, which was installed with MOLEKULAR. It will connect your computer to the Internet and activate your MOLEKULAR installation. In order to activate your copy of MOLEKULAR, you have to perform the following steps within the Service Center:

Log in: Enter your Native Instruments user account name and password on the initial page. This is the same account information you used in the Native Instruments Online Shop, where you bought your REAKTOR Instrument, and for other Native Instruments product activations.

Select products: The Service Center detects all products that have not yet been activated and lists them. You can activate multiple products at once—for example, several REAKTOR Instruments.

Activate: After proceeding to the next page, the Service Center connects to the Native Instruments server and activates your products.

Download updates: When the server has confirmed the activation, the Service Center automatically displays the Update Manager with a list of all available updates for your installed products. Please make sure that you always use the latest version of your Native Instruments products to ensure they function correctly.



Downloading updates is optional. After activation is complete, you can always quit the Service Center.

3 How to Use MOLEKULAR

The following sections will give you a brief overview over some basic operations: you will learn how to open MOLEKULAR, how to explore the factory-set Snapshots and how to load and play MOLEKULAR Snapshots from the Header and the Sidepane.



For latest information on REAKTOR PLAYER files and using Snapshots please refer to the REAKTOR Getting Started Guide.

As MOLEKULAR is an effect unit—albeit an immensely potent one—you will first need to load a sound source. This can be e.g. a sampler or a synthesizer. To do so, follow the common procedure of your sequencer software.



For many of MOLEKULAR's features to work as intended, MIDI clock needs to be running. We therefore recommend you use MOLEKULAR together with host software.

- ▶ To load REAKTOR or REAKTOR PLAYER into an effect slot in your sequencer software, follow the common procedure of your sequencer software.

3.1 How to Open MOLEKULAR

This is how to open MOLEKULAR in REAKTOR or REAKTOR PLAYER:

1. Start REAKTOR or REAKTOR PLAYER respectively.

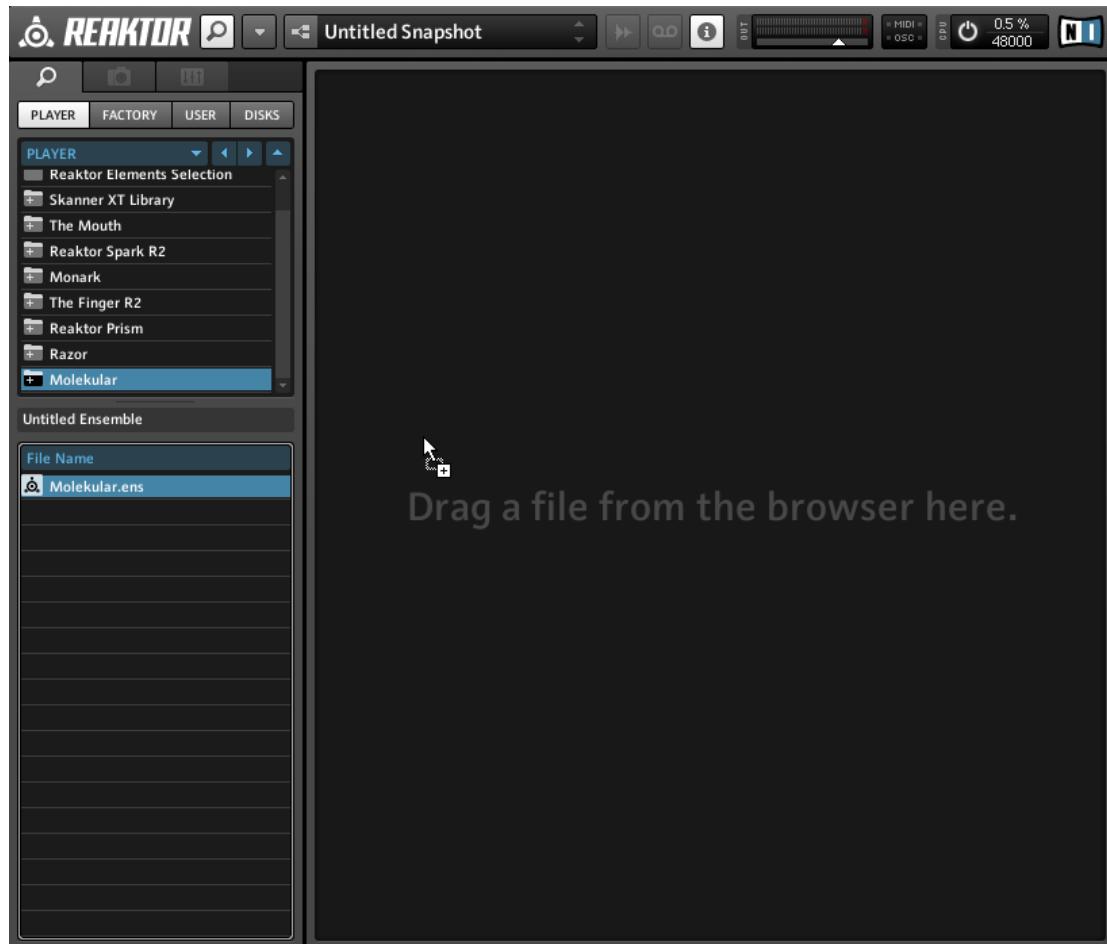
2. In the Browser on the left side of the REAKTOR / REAKTOR PLAYER window, click the [PLAYER](#) button to show the REAKTOR PLAYER files (you can open the browser with the [F5] key from your keyboard).



3. Click the [Molekular](#) folder. The content of the folder will be displayed in the lower section of the browser.



4. Double-click the [Molekular.ens](#) file, or drag it into the main screen.



5. MOLEKULAR will be loaded in REAKTOR / REAKTOR PLAYER:



3.2 Exploring Factory-set Snapshots

Record some notes in your DAW software and play them back to get an idea of how the Ensemble affects the source sound. Then, let's change the sound completely by loading a different Snapshot.



A Snapshot is REAKTOR's notion for a sound, preset, or patch. MOLEKULAR can hold banks of Snapshots, and loading any of these Snapshots will set each control of that Instrument to a specific value, and re-create a particular sound.

The Snapshots of MOLEKULAR are accessible from the central control in REAKTOR PLAYER's Header (Main Bar) or from the Sidepane.



The MOLEKULAR interface with Snapshot list in the Sidepane.

- (1) Sidepane button
- (2) Snapshot drop-down menu
- (3) Snapshot tab
- (4) Snapshot Banks
- (5) Snapshots

3.2.1 Loading a Snapshot from the Sidepane

If not already visible after startup, you need to open the Sidepane. The Sidepane holds a full overview of REAKTOR's Snapshot Banks and Snapshots from the currently selected Snapshot Bank.

1. Click the Sidepane button (1) in the Header to open the Sidepane.
2. Click the Snapshot tab (3) to access Snapshots.
3. Select a Snapshot Bank (4).
4. Select the name of a Snapshot entry (5).

→ The name of the selected Snapshot will be highlighted in the Sidepane, and the Snapshot loaded and ready in MOLEKULAR.

3.2.2 Loading a Snapshot from the Header

Loading a Snapshot from the REAKTOR PLAYER drop-down menu in the Header is the simplest way to interact with Snapshots.

1. Click the Snapshot drop-down menu (2). The menu holds all Snapshots and Banks of the Instrument.
2. Click an entry to select it.

3.3 Saving a Snapshot

Snapshots can only be saved when using the full version of REAKTOR, however, all your settings will be recalled perfectly in a host if you are using REAKTOR PLAYER, so you can tweak a sound perfectly for your song. All parameter settings made in MOLEKULAR will be saved as part of your DAW project and/or as a REAKTOR Preset. Please read the REAKTOR documentation for more information on plug-in mode.



For the latest information on REAKTOR PLAYER please refer to the REAKTOR Getting Started Guide.

3.4 Selecting MOLEKULAR A and B Panel Views

REAKTOR allows for each Instrument to have two separate panel layouts, A and B. You can switch between the A and B panel views by right-clicking on an empty space of MOLEKULAR's interface and clicking the appearing *View A* or *View B* context menu entry. For more information about the two views, see section [↑4, Overview of the MOLEKULAR Interface](#).

View A



MOLEKULAR View A

View B



MOLEKULAR View B

4 Overview of the MOLEKULAR Interface

MOLEKULAR's two separate panel views each serves its own purpose. In View B—the performance view—you gain access to the most important controls for playing live and to quickly process the input signal to maximum effect. View A—the additional view—on the other hand, gives you full access to the entire MOLEKULAR interface with all its audio processing and modulation possibilities. You can meticulously edit sound, modulate and cross modulate parameters and alter the routing of the semi modular system to your own liking.

4.1 View B Controls

The performance view is easy to navigate and gives you access to the most crucial controls for playing live and lets you morph between sounds using the powerful morpher.



View B

- (1) **Input**: Turn to adjust the level of the input signal.
- (2) **Mod** (global): Turn to set the amount of overall modulation. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).
- (3) **Output**: Turn to adjust the level of the output signal.
- (4) **Dry**: Turn to set the amount of dry signal to output.
- (5) **Motion**: Activate to modulate the morpher. For detailed information, see section [7.3, Morph Modulation: Motion](#).
- (6) **Morpher**: Click and drag the cursor to morph between five preset sounds (center, A, B, C, and D). For detailed information, see section [7, Morphing](#).

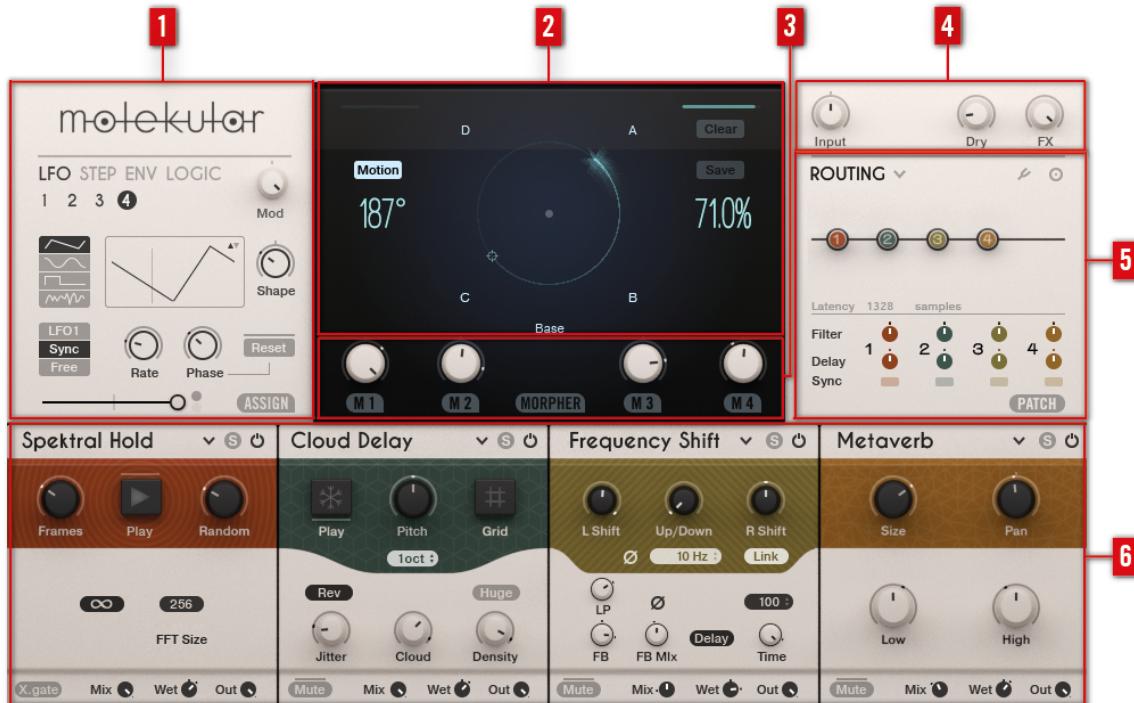
(7) Macro controls: Turn any of the four controls to adjust multiple settings simultaneously. For detailed information, see section [16, Macro Controls](#).



Press [Shift] on your computer keyboard while clicking and dragging a parameter to adjust the parameter in finer increments.

4.2 View A Sections

These are the main sections of the additional view. Each section gives you access to immense possibilities of altering the audio in creative ways.



View A

(1) Modulation section: Use this section to set up modulation for many of MOLEKULAR's parameters. It allows you to assign four LFOs, four step sequencers, three envelope generators and one envelope follower, and four logic units to modulate the parameters of the DSPs and the morpher as well as each other! For detailed information, see section [15, Modulation](#).

(2) Morpher (and center display): Use this section to save sound variations and morph between them with the mouse cursor or via modulation. The term here used for morph modulation is **Motion**. For detailed information, see section [17, Morphing](#). The center display is also used for the modulation screen that lets you observe and adjust modulation sources for any assigned destination. For detailed information, see section [15.3, Managing Modulation via the Modulation Screen](#). Furthermore, the pitch quantizer and some of the DSPs make use of pattern editors, which also are displayed at the center of the interface.

(3) Macro controls: Use this section to assign parameters and control them via macros. For detailed information, see section [16, Macro Controls](#).

(4) Master controls: Use this section to adjust levels of input and amount of audio processing. These controls are Snapshot independent and cannot be modulated. For details, see section [18, Routing, Patching, and Pitch Quantization](#).

(5) Routing section: Use this section to access controls for the routing system—including flowcharts and patching, as well as additional filters and delays—and for the powerful pitch quantization and morph quantization pages. For detailed information, see sections [18, Routing, Patching, and Pitch Quantization](#) and [17.4, Morph Quantization](#).

(6) DSPs (digital sound processors): Use the four DSPs—each with up to 14 different effect units—to process the input audio in interesting and creative ways. The selectable effect units range from traditional filters and reverb units to complex samplers and new creative effects. For detailed information, see section [19, DSP Reference](#).

5 Modulation

Many parameters in the MOLEKULAR interface can be modulated. To make use of the vast modulation possibilities, refer to the upper left section of the MOLEKULAR interface. Here you can assign and record modulation as well as adjust the parameters of each modulation source. For detailed information on all available parameters, see section [5.7, Modulation Unit Reference](#).



The modulation section with LFO 1 selected

Use the sections below to learn about workflows involving modulation. Remember that modulating some parameters can alter the output sound in unexpected ways and that it is important to save your work regularly.

5.1 Assigning Modulation

To assign modulation to e.g. a DSP parameter:

1. To select the modulation type, click the corresponding title ([LFO](#), [STEP](#), [ENV](#), or [LOGIC](#)).



2. To select one of the four available units, click the corresponding number, e.g., [1](#).



3. To enter assignment mode, click the **ASSIGN** button in the bottom right corner of the modulation section.



A gray slider—the modulation depth slider—appears next to all parameters available for modulation.



The DSPs must be switched on for their parameters to be visible.

4. Click and drag the bipolar slider in either direction to assign and set the modulation depth for the corresponding parameter.



The parameter is now assigned as modulation destination.

5. Repeat the previous step for as many parameters as you like.

6. To exit assignment mode, click the **ASSIGN** button one more time.



While in assignment mode, double-clicking any parameter's label opens the modulation screen for the corresponding parameter.

5.2 Assigning Cross Modulation



For information about modulating the morpher, see section [17.3, Morph Modulation: Motion](#).

You can also modulate the parameters of other modulators. To do so:

1. To select the modulation type, click the corresponding title ([LFO](#), [STEP](#), [ENV](#), or [LOGIC](#)).



2. To select one of the four available units, click the corresponding number, e.g., [1](#).



3. To enter assignment mode, click the **ASSIGN** button in the bottom right corner of the modulation section.



A key lock icon will appear next to the **ASSIGN** button.

4. To lock assignment mode to the selected modulator, click the key lock icon.



The **ASSIGN** button switches to indicate that assignment mode is now locked to the selected modulator.

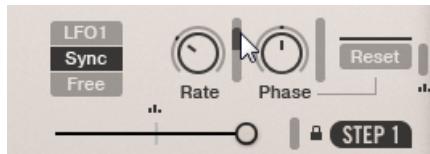


5. While in locked assignment mode, select the modulator you wish to modulate by clicking either another number from the same modulation type (e.g. **STEP 2**, **3**, or **4**) or another modulation type altogether (e.g. **LFO**, **ENV**, or **LOGIC**).

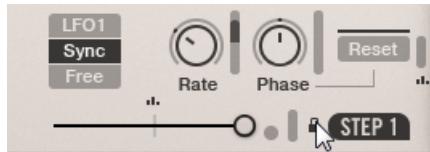


A gray slider—the modulation depth slider—appears next to all parameters available for modulation.

6. Click and drag the bipolar slider in either direction to assign and set the modulation depth for the corresponding parameter.

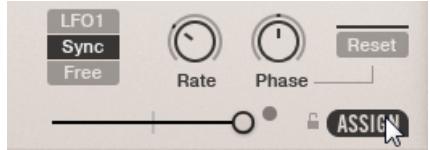


7. Repeat the previous two steps for as many modulators and parameters as you like.
8. To exit locked assignment mode, deactivate the key lock icon by clicking it.



The key lock opens and you enter "normal" assignment mode.

9. To exit assignment mode, click the **ASSIGN** button one more time.





While in assignment mode, double-clicking any parameter's label opens the modulation screen for the corresponding parameter.

5.3 Managing Modulation via the Modulation Screen

To manage all modulation that has been assigned to any one destination, open the modulation screen. There you find sliders and level meters to give you control over all modulation sources. To access the modulation screen, follow the steps below.

1. To enter assignment mode, click the **ASSIGN** button in the bottom right corner of the modulation section.



A gray slider—the modulation depth slider—appears next to all parameters available for

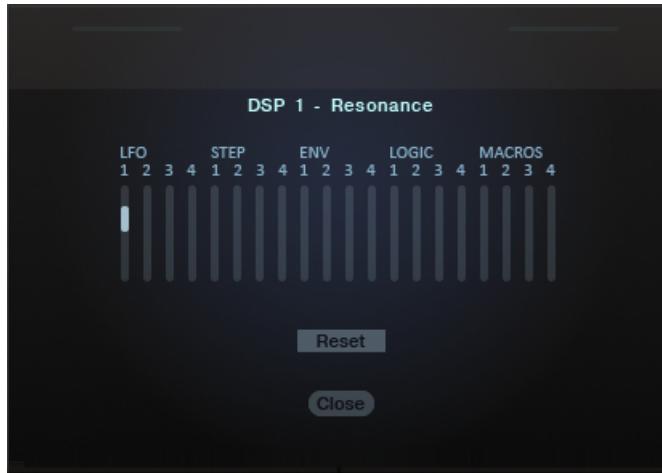
modulation. The slider will be darker for destinations for which modulation has been assigned.



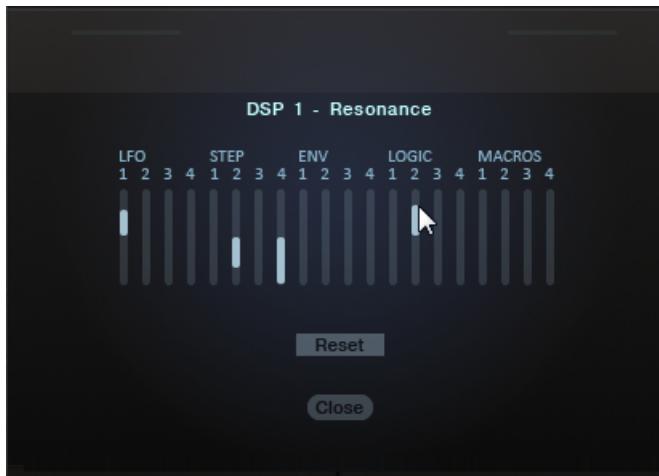
2. To open the modulation screen, while still in assignment mode, double-click the **label** of the parameter you want to manage. For parameters without labels, double-click the modulation screen icon (see section [15.3, Managing Modulation via the Modulation Screen](#)).



The modulation screen opens at the center of the interface.



3. Click and drag the sliders on the modulation screen to assign, adjust or remove modulation sources for the selected parameter.



You can also remove all assigned modulation sources from a parameter by clicking the Reset button in the modulation screen.

4. To exit the modulation screen, click [Close](#).

The Modulation Screen Icon

For parameters that are unlabeled—such as the [Mute](#) buttons in each DSP's footer or the two [Motion](#) controls of the morpher (angle and radius)—follow the steps below to open the modulation screen.

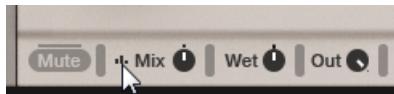
1. To enter assignment mode, click the **ASSIGN** button in the bottom right corner of the modulation section.



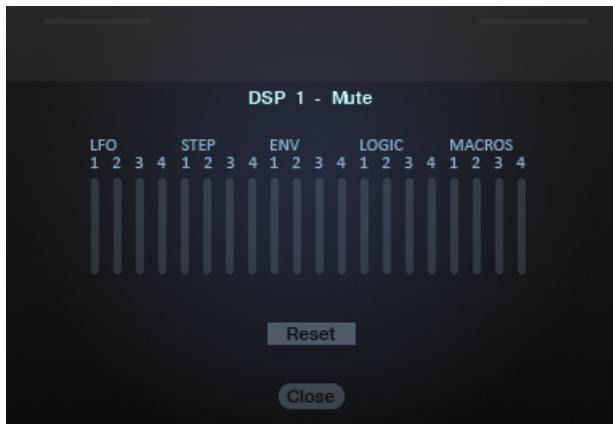
A gray slider—the modulation depth slider—appears next to all parameters available for modulation. The slider will be darker for destinations for which modulation has been assigned. Next to unlabeled parameters, e.g. the **Mute** button in the image below, the modulation screen icon also appears.



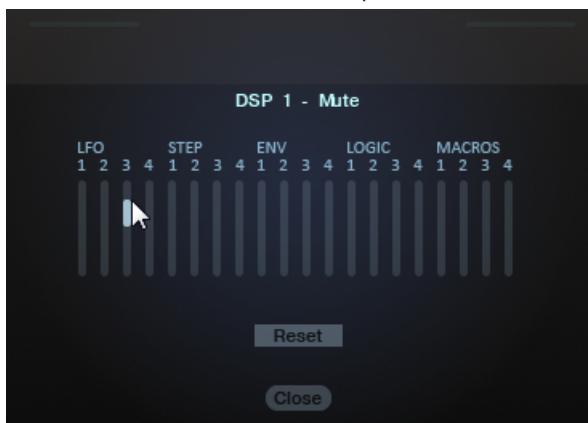
2. To open the modulation screen for an unlabeled parameter, double-click the modulation screen icon next to it.



The modulation screen opens at the center of the interface.



3. Click and drag the sliders on the modulation screen to assign, adjust or remove modulation sources for the selected parameter.



You can also remove all assigned modulation sources from a parameter by clicking the Reset button in the modulation screen.

4. To exit the modulation screen, click **Close**.

5.4 Removing Modulation Destinations

Besides removing modulation via the modulation screen (see section [5.3, Managing Modulation via the Modulation Screen](#)), there is another easy way to remove modulation from an assigned parameter. To delete assigned modulation and remove a modulation source:

1. To enter assignment mode, select the modulation source you wish to remove (e.g. [STEP 3](#)) and click the **ASSIGN** button in the bottom right corner of the modulation section.



A gray slider—the modulation depth slider—appears next to all parameters available for modulation. The slider will be darker for parameters for which modulation has been assigned.



2. To remove modulation, double-click the slider.



3. To exit assignment mode, click the **ASSIGN** button one more time.



While in assignment mode, double-clicking any parameter's label opens the modulation screen for the corresponding parameter.

5.5 Clearing All Modulation

Sometimes it can be necessary to clear all assigned modulation. Remember that doing so effectively removes all modulation, so remember to save your work first to be able to retrieve the current state of MOLEKULAR. To clear all modulation:

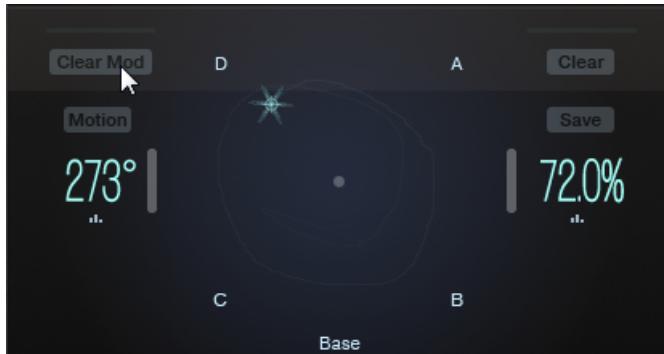
1. To enter assignment mode, click the **ASSIGN** button in the bottom right corner of the modulation section.



When you enter assignment mode, the **Clear Mod** button appears in the upper left corner of the center display.



2. To remove all assigned modulation, click the **Clear Mod** button.



→ All modulation assignments are removed.

5.6 Stereo Modulation

Stereo modulation gives you the possibility to split the modulation signal in two separate channels, whereof one is phase inverted. All parameters available for stereo modulation assignment display a stereo modulation icon next to them while in assignment mode.

1. To enter assignment mode, select the modulation source you wish to assign (e.g. **LFO 1**) and click the **ASSIGN** button in the bottom right corner of the modulation section.



A gray slider—the modulation depth slider—appears next to all parameters available for modulation. The slider will be darker for destinations for which modulation has been as-

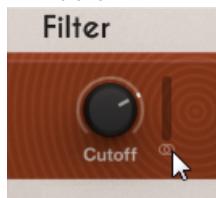
signed. The stereo modulation icon appears next to all parameters available for stereo modulation (i.e. [Cutoff](#) in the example below).



2. To assign the parameter as modulation destination, click and drag the corresponding modulation depth slider in either direction.



3. To apply stereo modulation, click the stereo modulation icon.



The modulation signal is split in two, whereof one is phase inverted.

4. To exit assignment mode, click the **ASSIGN** button one more time.



5.7 Modulation Unit Reference

In this section you will find detailed descriptions of all parameters of each modulation source.



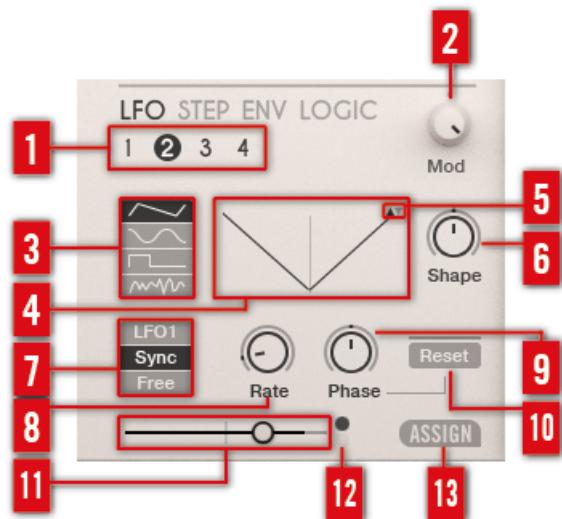
Press [Shift] on your computer keyboard while clicking and dragging a parameter to adjust the parameter in finer increments.

5.7.1 LFO

MOLEKULAR comes with four LFOs (low frequency oscillators) that each can be assigned to modulate any number of effect parameters as well as the morphing and macro controls and even the parameters of other modulation sources. You can sync the rate of LFOs **2**, **3**, and **4** to LFO **1**.



It is possible for an LFO to modulate the parameters of another LFO, but not its own controls. Thus, e.g. LFO 1 can modulate the parameters of LFO 2, 3, and 4, but not those of LFO 1.



LFO 2

(1) 1, 2, 3, and 4: Switch between the four available LFOs. Note that these are not on/off switches; the modulators are all active simultaneously.

(2) **Mod** (global): Turn this in either direction to adjust the amount of overall modulation. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).

(3) **Waveform**: Select one of four waveforms. Available for selection are **trisaw**, **sine**, **pulse**, and **noise**. The selected waveform is visible in the waveform display. Use the **Shape** control to further define each waveform.

(4) **Waveform display**: Use the display for visual feedback on the shape of the selected waveform.

(5) **Polarity**: Click to switch between unipolar and bipolar waveforms.

(6) **Shape**: Turn to adjust the shape of the selected waveform.

- With **trisaw** selected, turning **Shape** alters the waveform from uprising sawtooth to triangle at the center position and then to falling saw.

- With **sine wave** selected, turning **Shape** alters the waveform from spike wave to sine wave at the center position to square wave.
- With **pulse wave** selected, **Shape** acts as pulse-width control.
- With **noise** selected, turning **Shape** crossfades between time-wise random signal at fully left, via stepped random modulation at center position to smooth-edged stepped random modulation at fully right.

(7) **Rate mode:** Select the mode for the **Rate** control.

- **LFO 1** (only available for LFO 2, LFO 3, and LFO 4): Select this to sync to LFO 1. In this mode, **Rate** is set as multiples of LFO 1's rate.
- **Sync**: Select this to sync to the tempo of the host software.
- **Free**: Select this to set the rate of the LFO in Hz using the **Rate** control.

(8) **Rate:** Turn to control the speed of the LFO's fluctuations. Use the rate mode selector to select the unit of the **Rate** parameter.

(9) **Phase:** Turn to control the phase offset and adjust the start position for phase reset. The current position of the **Phase** control is visually reflected in the waveform display.

(10) **Reset:** Click to restart the LFO at the phase position set with **Phase**.

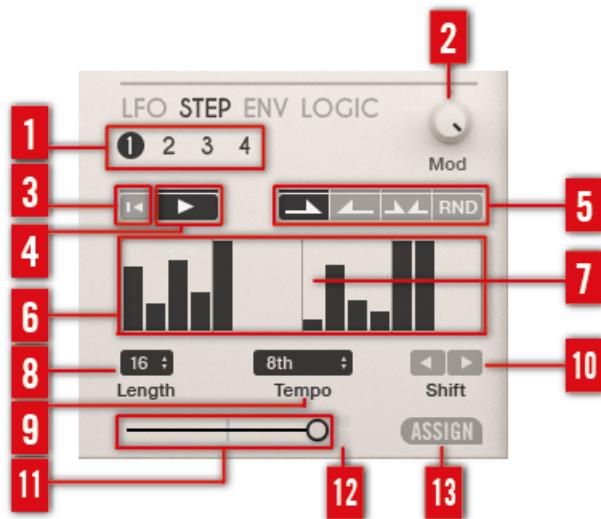
(11) **Modulation depth:** Slide this in either direction to set the modulation depth for the corresponding modulator. The amount is visually reflected by the modulation depth indicator. Combine the setting of the modulation depth with the overall **Mod** control to set the desired modulation depth.

(12) **Modulation indicators:** These indicators blink to visualize the set modulation depth. High **Rate** settings and large values for modulation depth equals faster blinking. When the polarity button is set to unipolar, only one of the indicators blink; when set to bipolar, they both blink.

(13) **ASSIGN:** Activate to enter assignment mode. When active, you can assign modulation for any parameter that displays a gray slider. For detailed information on assigning modulation, see section [↑5.1, Assigning Modulation](#).

5.7.2 Step Sequencer

Four identical step sequencers are available and can be assigned to modulate any number of effect parameters as well as the morphing and macro controls and even the parameters of other modulation sources.



STEP 1

(1) **1, 2, 3**, and **4**: Switch between the four available step sequencers. Note that these are not on/off switches; the modulators are all active simultaneously.

(2) **Mod** (global): Turn this in either direction to adjust the amount of overall modulation for all modulators simultaneously. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).

(3) **Reset**: Activate to set the start position to the first step of the sequence. Deactivate it to start from the current position of the sequence whenever you click the play button.

(4) **Play**: Click to start and stop playing the sequence.

(5) **Direction**: Select one of four modes to set the playing direction of the step sequence.

- **Forward**: Select this to play the sequence from left to right, i.e. step 1 - 16.

- **Backward:** Select this to play the sequence from right to left, .i.e. step 16 - 1.
- **Pendulum:** Select this to play the sequence from left to right and then from right to left again, .i.e. step 1 - 16 - 1.
- **RND:** Select this to play the sequence in random order, .i.e. step X - Y.

(6) Steps: Click and drag the sliders to set values for the corresponding steps 1 - 16. The value of each step represents the modulation depth for that step in the sequence, with larger values leading to stronger modulation. When using bipolar steps, the assigned parameter will be modulated in both directions from its current position.

(7) Step editor: Right-click in the step area to open the step editor. Use the controls to e.g. copy and paste sequences from one step sequencer to another.

- **Copy:** Click to place the displayed steps in the clipboard.
- **Paste:** Click to paste step data placed in the clipboard. Note that you can paste the copied values into any of MOLEKULAR's available sequencers, i.e. even across modulation types.
- **Random:** Click to randomize the steps.
- **Polarity** (small triangular icons): Click to switch between unipolar and bipolar settings for the steps. Use bipolar steps to modulate the assigned parameter in both directions from its current position.

(8) Length: Click and drag to select the amount of steps to be played in the sequence.

(9) Tempo: Click and drag to set the rate of the sequence's stepping in note values. The tempo is always synced to host tempo.

(10) Shift: Click the arrows to shift the step data one step in either direction.

(11) Modulation depth: Slide this in either direction to set the modulation depth for the corresponding modulator. The amount is visually reflected by the modulation depth indicator. Combine the setting of the modulation depth with the overall **Mod** control to set the desired modulation depth.

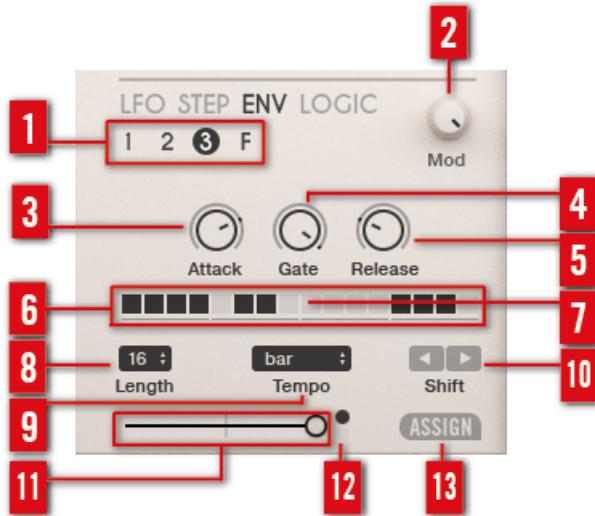
(12) Modulation indicators: These indicators blink to visualize the set modulation depth. High **Tempo** settings and large values for modulation depth equals faster blinking. When the polarity button is set to unipolar, only one of the indicators blink; when set to bipolar, they both blink.

(13) **ASSIGN**: Activate to enter assignment mode. When active, you can assign modulation for any parameter that displays a gray slider. For detailed information on assigning modulation, see section [5.1, Assigning Modulation](#).

5.7.3 Envelope

MOLEKULAR comes equipped with three envelope generators and sequencers as well as one envelope follower. Use the parameters of the envelope generators to set the shape of the envelope and the sequencer to trigger the gate. Select the envelope follower to measure the signal of the input, the output, or one of the effect units and use the signal as an envelope.

Envelope Generators and Sequencers



ENV 3

(1) 1, 2, 3, and F: Switch between the four available envelopes. Tabs 1, 2, and 3 open the corresponding envelope generators and their sequencers, while F opens the envelope follower. Note that these are not on/off switches; the modulators are all active simultaneously.

(2) Mod (global): Turn this in either direction to adjust the amount of overall modulation for all modulators simultaneously. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).

(3) Attack: Turn to set the duration of the envelope's attack phase. The behavior of the **Attack** parameter is largely dependent on the **Gate** setting.

(4) Gate: Turn to set the gate value for the steps in the sequencer. At a value of 50%, **Gate** stops the **Attack** phase of the envelope once 50% of the step length and starts the **Release** phase. Once the next step in the sequence is triggered, rather than beginning from zero position, the **Attack** is triggered at the envelope's current position, which leads to an escalating effect. At a value of 100%, the **Gate** instead lets the **Attack** time play out in its entirety.

(5) Release: Turn to set the duration of the envelope's release phase.

(6) Steps: Click the steps to trigger modulation of the assigned parameter during the corresponding phase in the sequence.

(7) Step editor: Right-click in the step area to open the step editor. Use the controls to e.g. copy and paste sequences from one envelope sequencer to another.

- **Copy:** Click to place the displayed steps in the clipboard.
- **Paste:** Click to paste step data placed in the clipboard.
- **Rndm:** Click to randomize the steps.

(8) Length: Click and drag to select the amount of steps to be played in the sequence.

(9) Tempo: Click and drag to set the rate of the sequence's stepping in note values. The tempo is always synced to host tempo.

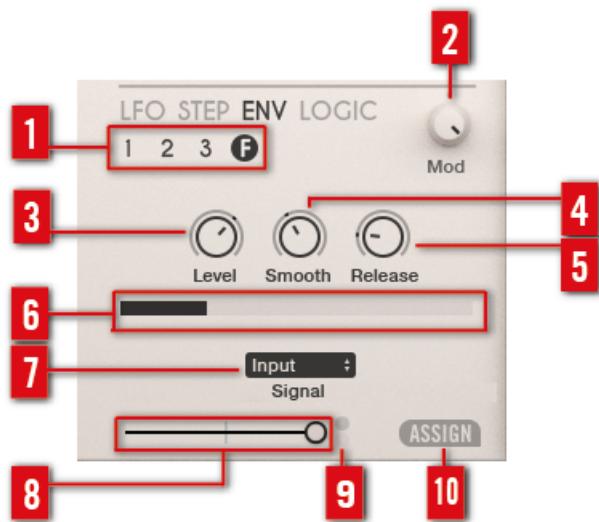
(10) Shift: Click the arrows to shift the step data one step in either direction.

(11) Modulation depth: Slide this in either direction to set the modulation depth for the corresponding modulator. The amount is visually reflected by the modulation depth indicator. Combine the setting of the modulation depth with the overall **Mod** control to set the desired modulation depth.

(12) Modulation indicators: These indicators blink to visualize the set modulation depth. High **Tempo** settings and large values for modulation depth equals faster blinking. When the polarity button is set to unipolar, only one of the indicators blink; when set to bipolar, they both blink.

(13) **ASSIGN**: Activate to enter assignment mode. When active, you can assign modulation for any parameter that displays a gray slider. For detailed information on assigning modulation, see section [↑5.1, Assigning Modulation](#).

Envelope Follower



ENV F

(1) **1**, **2**, **3**, and **F**: Switch between the four available envelopes. Tabs **1**, **2**, and **3** open the corresponding envelope generators and their sequencers, while **F** opens the envelope follower. Note that these are not on/off switches; the modulators are all active simultaneously.

(2) **Mod** (global): Turn this in either direction to adjust the amount of overall modulation for all modulators simultaneously. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).

(3) **Level**: Turn to adjust the level of the input signal.

(4) **Smooth**: Turn to smooth the reaction time of the follower. Use it to add some attack time and soften the initial phase of movements.

(5) **Release**: Turn to set the duration of the envelope's release phase.

(6) Level meter: This level meter displays the envelope follower's output signal.

(7) Signal: Click and drag to select the input signal. Available for selection are *Output*, *DSP 1*, *DSP 2*, *DSP 3*, *DSP 4*, and *Input*.

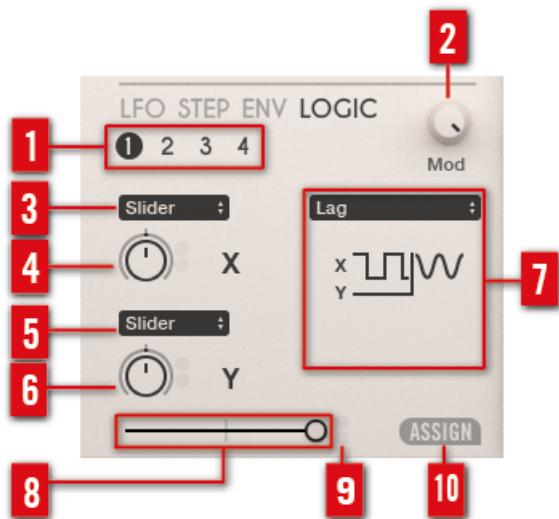
(8) Modulation depth: Slide this in either direction to set the modulation depth for the corresponding modulator. The amount is visually reflected by the modulation depth indicator. Combine the setting of the modulation depth with the overall *Mod* control to set the desired modulation depth.

(9) Modulation indicators: These indicators blink to visualize the set modulation depth. The input signal and modulation-depth values influence the blinking. When the polarity button is set to unipolar, only one of the indicators blink; when set to bipolar, they both blink.

(10) ASSIGN: Activate to enter assignment mode. When active, you can assign modulation for any parameter that displays a gray slider. For detailed information on assigning modulation, see section [↑5.1, Assigning Modulation](#).

5.7.4 Logic

There are four logic units that each has eight selectable modes in which they can operate. The logic units let you select two modulation sources as input signals. Use the logic units' eight available modes to create even more complex modulation by e.g. applying mathematical operations to the input modulation signals and then finally assigning the outgoing modulation signal to any destination you want. An example of using the *LOGIC* units is to let an LFO decide when a particular step in a step sequence should be triggered and when not, which can add unexpectedness and groove to a rhythm. See below for a list of the available modes.



LOGIC 1 with Lag mode selected

(1) 1, 2, 3, and 4: Switch between the four available logic units. Note that these are not on/off switches; the modulators are all active simultaneously.

(2) **Mod** (global): Turn this in either direction to adjust the amount of overall modulation for all modulators simultaneously. For optimized effect, leave this at maximum value (fully right; or fully left for phase inverted modulation).

(3) **Input X**: Click and drag to select a modulator as input. Select *Slider* to use the rotary control to set a static value.

(4) **Rotary control X**: Turn to set the modulation amount of the selected input. In some logic modes, the rotary controls instead set trigger thresholds. For details on each mode, see the list below.

(5) **Input Y**: Click and drag to select a modulator as input. Select *Slider* to use the rotary control to set a static value.

(6) **Rotary control Y**: Turn to set the modulation amount of the selected input. In some logic modes, the rotary controls instead set trigger thresholds. For details on each mode, see the list below.

(7) Mode: Click and drag the selector to select one of eight modes for the logic unit. A visual representation of the function is available below the selector in certain modes. For detailed information about each mode, see the list below.

(8) Modulation depth: Slide this in either direction to set the modulation depth for the corresponding modulator. The amount is visually reflected by the modulation depth indicator. Combine the setting of the modulation depth with the overall **Mod** control to set the desired modulation depth.

(9) Modulation indicators: These indicators blink to visualize the set modulation depth. The selected input signals and modulation depth values influence the blinking. When the polarity button is set to unipolar, only one of the indicators blink; when set to bipolar, they both blink.

(10) ASSIGN: Activate to enter assignment mode. When active, you can record modulation for any parameter that displays a gray slider. For detailed information on assigning modulation, see section [15.1, Assigning Modulation](#).

Step Mem.



Step Mem.

In the *Step Mem.* mode, you have access to the buffers of the four step sequencers found in the **STEP** tab, which are represented here by the **1|2|3|4** selector (7). Use the **X** input (3) to define a particular step in the sequence and the **Y** input (5) to decide whether it should be triggered or not. Rotary control **Y** (6) here sets the level of the threshold. You can flip the roles of the two input signals by clicking the **X Y** visualization (7) below the mode selector. Note that by doing so you also swap the corresponding functions of the rotary controls.

(7) [1](#)[2](#)[3](#)[4](#): Select the step sequencer you want to control.

Random



Random

In *Random* mode, the logic unit uses an internal random generator as source for S/H (sample and hold). Use the **X** input (3) to define the amount of possible outcomes, i.e. randomness ([Resolution](#)), and the **Y** input (5) as sampling trigger ([Trigger](#)). Turn the bipolar rotary control **X** (4) fully right or fully left for full random scale, or set it to center position for a sample rate of two. Rotary control **Y** (6) sets the level of the threshold. You can flip the roles of the two input signals by clicking the **X Y** visualization (7) below the mode selector. Note that by doing so you also swap the corresponding functions of the rotary controls.

(7) **Smooth**: Turn to smooth the signal between the sample points.

Sample & Hold



Sample & Hold

The *Sample & Hold* mode uses one signal to trigger sampling of the other, which is then held until the next sample trigger. Rotary control **Y** (6) sets the level of the trigger threshold. You can flip the roles of the two input signals by clicking the **X Y** visualization (7) below the mode selector. Note that by doing so you also swap the corresponding functions of the rotary controls.

Lag



Lag

Lag mode allows you to smooth the signal at input **X** (3) by an amount set by input **X** (6). You can flip the roles of the two input signals by clicking the **X Y** visualization (7) below the mode selector.

Gate



Gate

This logic gate uses Boolean logic to set on/off values. Whenever an input signal goes high, it gets the value 1; whenever an input signal goes low, it gets the value 0. Select one of three gates to decide when to trigger an output signal.

AND: Select this to output the signal when both **X** and **Y** are high simultaneously.

OR: Select this to output the signal when either **X** or **Y** (or both!) is high.

XOR (exclusive or): Select this to output the signal when and only when either **X** or **Y** (not both!) is high.

N: Activate to negate the result of the operation, turning 1 to 0 and vice versa.

Math



Math

The *Math* mode lets you apply mathematical operations to the input signals in order to decide the output.

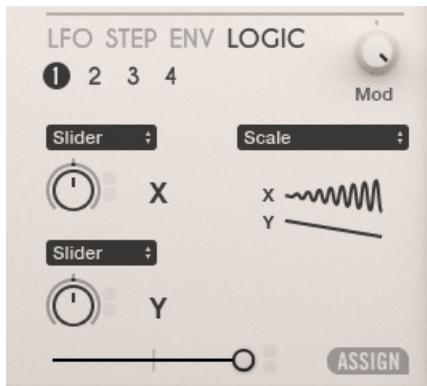
SUM: Select this to add **X** and **Y** together and use the sum as output signal.

MAX: Select this to measure the levels of **X** and **Y** and only use the input signal with the **largest** value as output signal.

MIN: Select this to measure the levels of **X** and **Y** and only use the input signal with the **smallest** value as output signal.

AVG: Select this to measure the levels of **X** and **Y** and calculate the average of both signals. The average value will be used as output signal.

Scale



Scale

In *Scale* mode, you can use one input signal to scale the other input. For example, setting one input selector to *Slider* and the other to e.g. an LFO allows you to multiply the LFO curve by the factor set with the scale control. You can flip the roles of the two input signals by clicking the **X Y** visualization (7) below the mode selector.

X-Fade



X-Fade

X-Fade mode gives you access to a simple crossfader that lets you mix between the two input signals.

(7) Crossfader: Slide this fully down to use only input **Y** for modulation, or fully up to use only input **X** for modulation. Mix between the two to use both signals.

6 Macro Controls

Beyond the modulation section, MOLEKULAR also gives you the possibility to assign four macro controls—located at the center of the interface (and also available in Performance view!)—to an unlimited amount of parameters. In that way you can control multiple parameters simultaneously by the turn of a knob. The macro controls can furthermore be automated from your host software, giving you full control of MOLEKULAR with just the turn of a single control.



The macro controls

(1) Macro controls: Turn the corresponding macro control to adjust the modulation depth of all assigned parameters.

(2) **M 1, M 2, M 3, and M 4 (macro labels):** Click to enter macro assignment mode for the corresponding macro control. Note that you cannot cross assign macro controls to each other.

Assigning Macros

Macro assignment is done much in the same way as modulation assignment. For detailed information on the modulation section, see section [15, Modulation](#). You can assign the macro controls to control parameters in the modulation section, but you cannot assign them to other macros or the morpher.

1. To select a macro control and enter assignment mode, click its label (M 1, M 2, M 3, or M 4).



A gray slider—the modulation depth slider—will appear next to all parameters available for macro control.

2. Click and drag the bipolar slider in either direction to assign modulation for the corresponding parameter.



3. Repeat the previous step for as many parameters as you like. You can assign an unlimited amount of parameters to each macro control.



4. To exit assignment mode, click the macro label one more time.



7 Morphing

At the center of the MOLEKULAR interface you find the powerful **MORPHER**. Use it to dramatically alter the sound either by clicking and dragging your mouse or by modulating the angle and radius controls, located on each side of the center display.

7.1 The Morph Controls



The morph controls

- (1) **A, B, C, D, and Base**: These are morph slots, where you can save the current state of MOLEKULAR and morph between the saved states. For detailed information on how to assign the slots, see section [7.2, Morph Assignment](#).
- (2) **Motion**: Activate to modulate the angle and radius controls with one or more assigned modulation source(s). For details, see section [7.3, Morph Modulation: Motion](#).
- (3) **Angle**: Click and drag to set the position of the cursor.

(4) Radius: Click and drag to set the position of the cursor, where 0.0% is center position (the morph base) and 100.0% is represented by the circumference.

(5) Cursor: This shows the current position of the morpher. Click and drag it to morph between the offsets, and/or modulate its movements by activating [Motion](#).

(6) Clear: Click to enter morph clearing mode. For details, see sections [17.2.3, Removing Morph Base or Offsets](#) and [17.2.4, Removing Morph Destinations](#).

(7) Save: Click to enter morph saving mode. For details, see section [17.2.2, Assigning a Morph Base and Offsets](#).

(8) MORPHER: Click to enter morph assignment mode, which lets you select what DSPs and modulators should be available for morphing.

7.2 Morph Assignment

To assign and modulate the morpher follow the steps below.

7.2.1 Selecting Available Destinations

To make use of the morpher, you must first select which DSPs and modulators to morph.

1. To assign the morpher, click the **MORPHER** button.



All units available for morphing are labeled either **ON** or **OFF** (or as a black circle around the modulator number).



2. To select a unit for morphing, set the switch to **ON**.



3. Repeat for as many units as you want to control.
4. To exit assignment mode, click the **MORPHER** button one more time.



7.2.2 Assigning a Morph Base and Offsets



To assign morph base and offsets, you must first select what units to include in the morpher. For details, see section [7.2.1, Selecting Available Destinations](#).

First you have to save a morph base. Without [Base](#), you cannot yet save offsets. To save the current state of MOLEKULAR as the morph base:

1. To enter morph saving mode, click [Save](#).



[Save](#) is now blinking to indicate morph saving mode.

2. To save the current state as the morph base, click [Base](#).



The morph base is now saved and is represented at the center of the morpher.

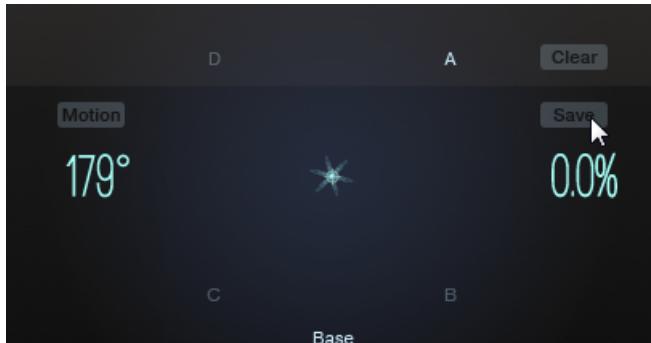


3. Play around with the morpher by clicking and dragging the cursor!

Assigning Offsets

To save offsets to the base and save them in the four slots, follow the steps below.

1. To save an offset to the morph base, enter morph saving mode by clicking [Save](#).

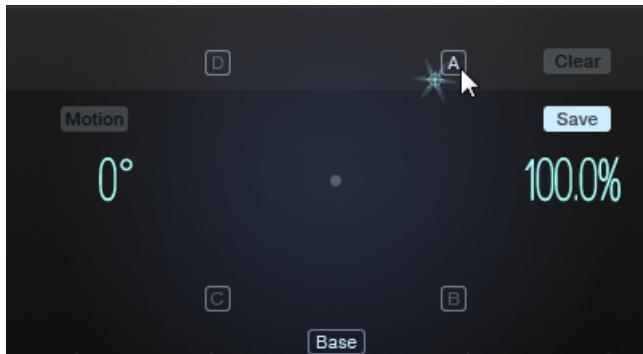


[Save](#) is now blinking to indicate morph saving mode.

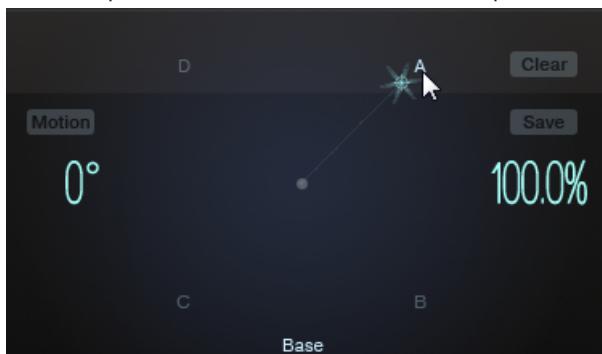
2. Adjust the parameters of the effect units and modulators you want to morph accordingly.



3. To save the current state of MOLEKULAR as a morph offset, click one of the four letters (A, B, C, or D).

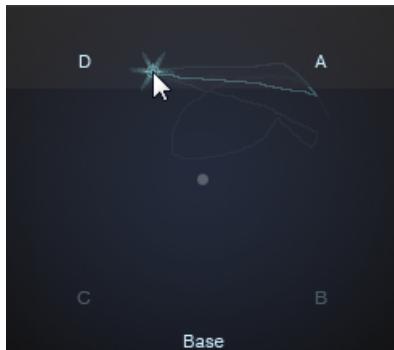


The morph offset is now saved and is represented by the corresponding letter.



4. To save offsets for the remaining slots, repeat steps 3 to 5 but select a different letter each time.

5. To morph between the base and its offsets, click and drag the cursor from one point to another or modulate via Motion (see section [17.3, Morph Modulation: Motion](#)).



- ▶ The sound morphs from one saved state to another and the morphing is visually represented by the movement of the cursor.



For detailed information about how to define the rate at which the morpher moves from one sound to another, see section [17.4, Morph Quantization](#).

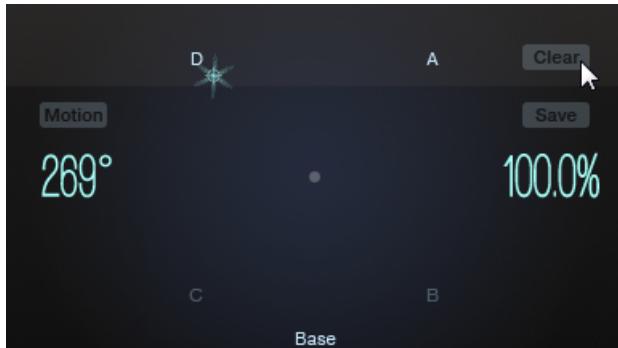
7.2.3 Removing Morph Base or Offsets

If you are unhappy with one or more of the offsets you saved, you can easily remove these.

Clearing an Offset

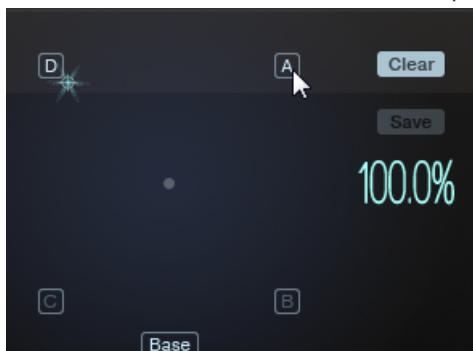
To remove an offset:

1. To enter morph clearing mode, click Clear.



Clear is now blinking to indicate morph clearing mode.

2. To remove an offset, click the corresponding letter.



→ The slot is now empty and you can save a new offset for that letter.

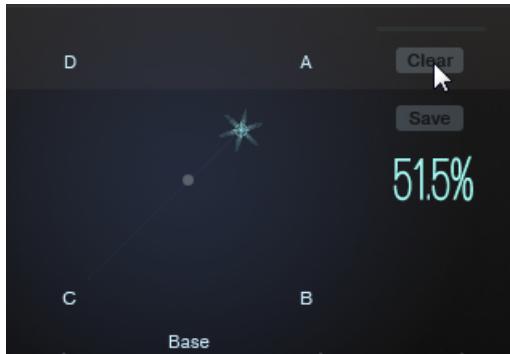
Clearing the Base



Clearing the Base effectively deletes all saved morph states, including the offsets.

To remove the base and reset the morpher completely:

1. To enter morph clearing mode, click **Clear**.



Clear is now blinking to indicate morph clearing mode.

2. To remove the base and clear all saved morphing, click **Base**.

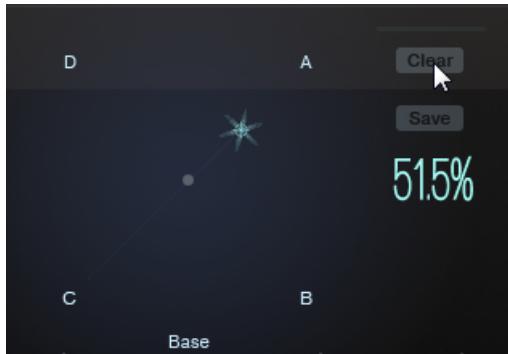


→ All slots are now empty and you can save a new base.

7.2.4 Removing Morph Destinations

If you assigned a parameter to the morpher and you are unhappy with the result, you can remove parameters one by one.

1. To enter morph clearing mode, click **Clear**.



Clear is now blinking to indicate morph clearing mode.

2. To clear morphing for one parameter, double-click the corresponding parameter **label**.



- The parameter has been removed from morphing.

7.3 Morph Modulation: Motion

As with many other features of MOLEKULAR, the morphing section can also be modulated. For detailed information about modulation, see section [15, Modulation](#).

1. Assign a morph base and offsets as described in section [17.2.2, Assigning a Morph Base and Offsets](#).

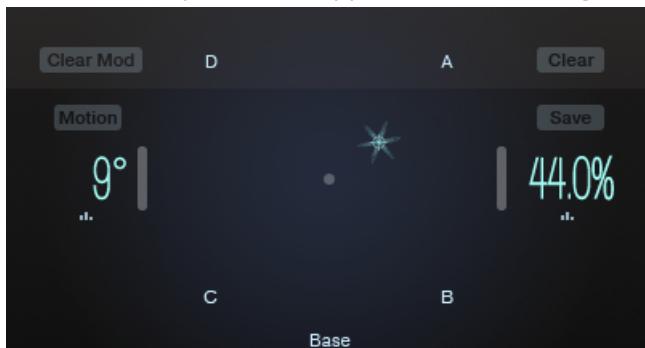
2. To select a modulation source, click the corresponding tab and number in the modulation section.



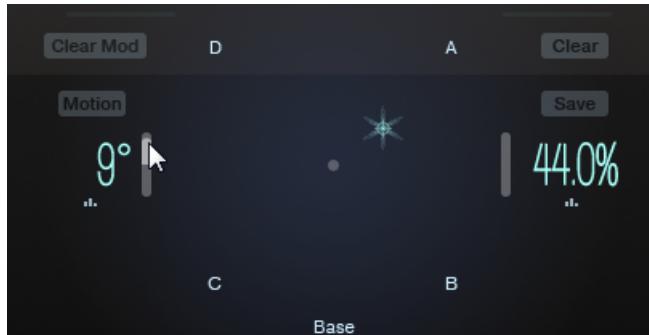
3. To enter assignment mode, click the **ASSIGN** button.



Modulation depth sliders appear next to the angle and radius controls.



4. Click and drag the bipolar sliders to record modulation.

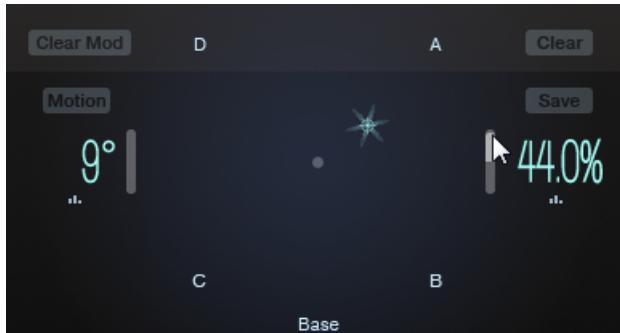


5. To use more than one modulation source, while still in assignment mode, select another modulator.



Modulation depth sliders corresponding to the newly selected modulator appear next to the angle and radius controls.

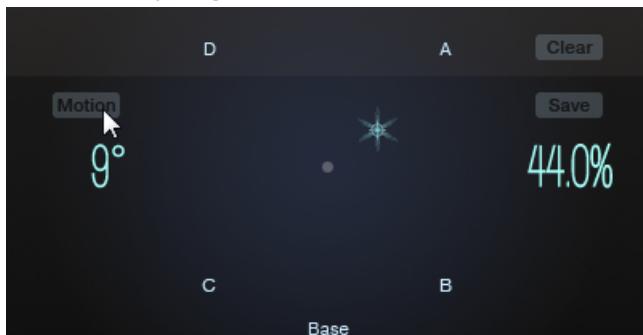
6. Click and drag the bipolar sliders to record modulation.



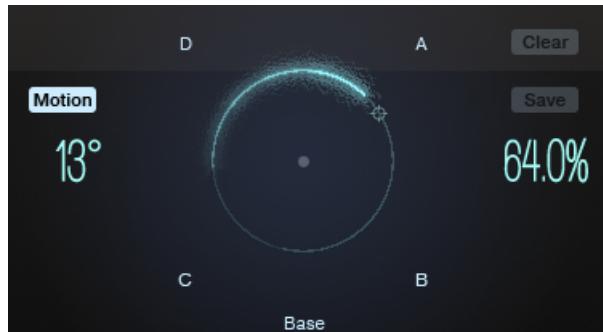
7. To exit assignment mode, click the **ASSIGN** button again.



8. To start morphing, click the **Motion** button.



- The morpher moves automatically between the offsets in accordance with the assigned modulation sources.



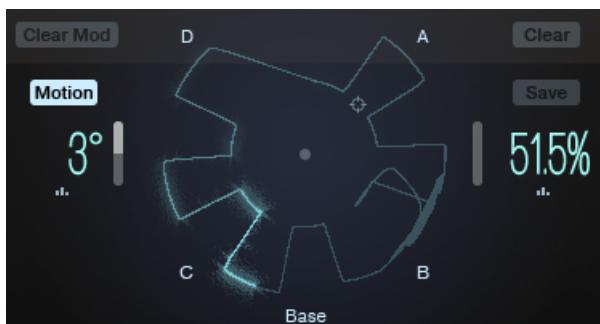
7.3.1 Managing Motion via the Morph Modulation Screen

To manage Motion, open the morph modulation screen:

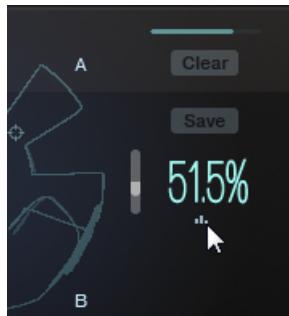
1. To enter assignment mode, click the **ASSIGN** button.



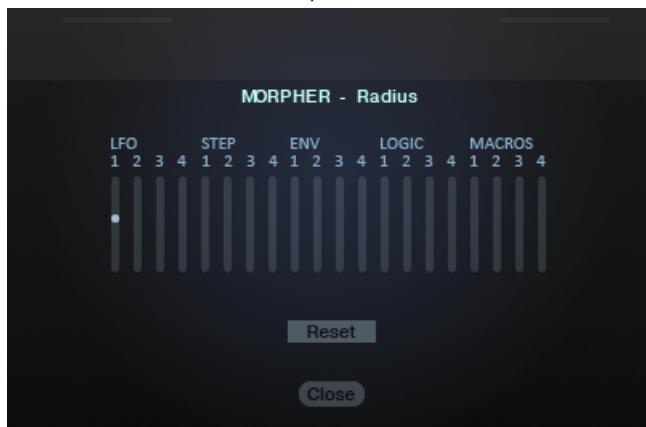
Modulation depth sliders as well as the modulation screen icon appear next to the angle and radius controls.



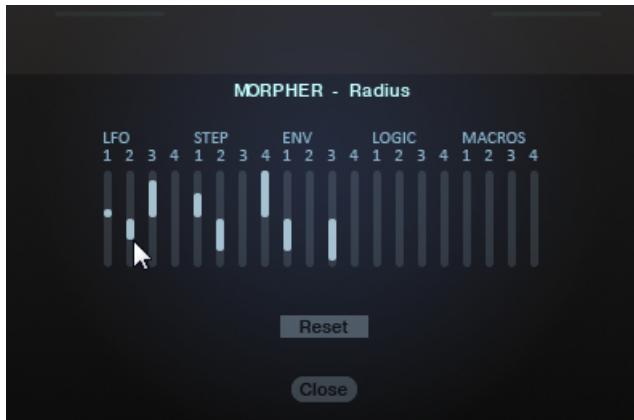
2. To open the modulation screen, double-click the modulation screen icon beneath either the angle or radius control.



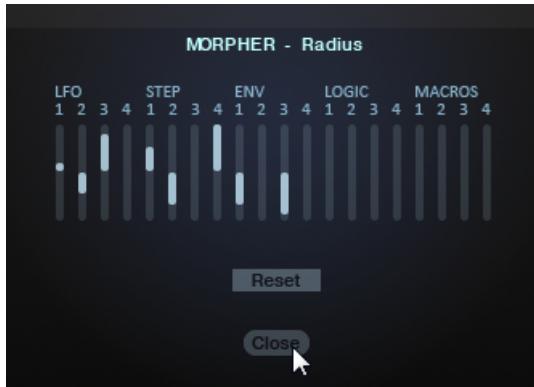
The modulation screen opens at the center of the interface.



3. To assign modulators to the selected Motion control (angle or radius), click and drag the corresponding sliders.



4. To close the modulation screen, click the [Close](#) button.



5. To exit assignment mode, click the **ASSIGN** button one more time.



7.4 Morph Quantization

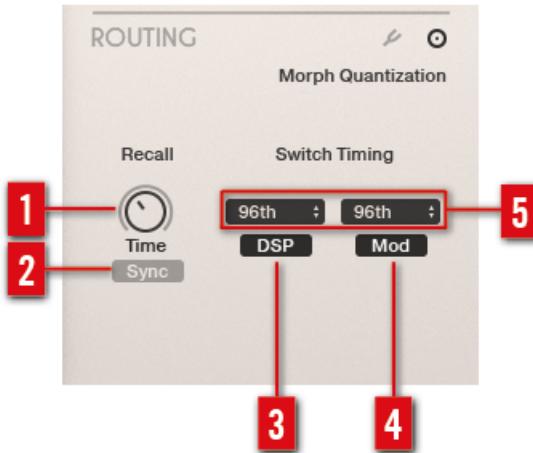
The **Morph Quantization** page is found near the upper right corner of the interface. Use it to gain precise control over the rate at which the morpher moves from one offset stage to another.

- To open the **Morph Quantization** page, click the morpher icon to the very right.



The Morph Quantization Page

The Morph Quantization page gives you access to the following controls:

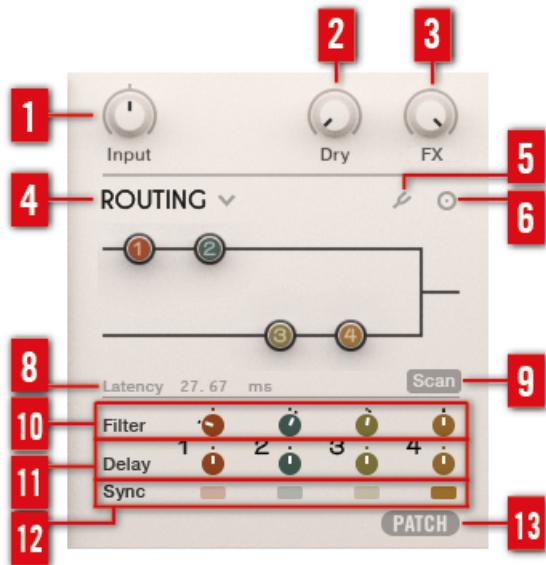


The Morph Quantization page

- (1) **Recall Time**: Turn to adjust the speed at which the morpher moves from one stored offset to another. Note that you have to first save a morph base and offsets for the morph quantization to work properly (see section [7.2, Morph Assignment](#)). Depending on the Sync setting, you can either set the time in seconds or in note values.
- (2) **Sync**: Activate to set the morph time in note values ranging from 1/96th to 32 bars. When Sync is inactive, you instead set the Recall Time control in seconds, ranging from 0.046 to 28.8.
- (3) **DSP**: Activate to apply Switch Timing to the DSPs. Only assigned morph destinations are affected.
- (4) **Mod**: Activate to apply Switch Timing to the modulators. Only assigned morph destinations are affected.
- (5) **Switch Timing**: Click and drag these selectors (left for DSPs; right for modulators) to set the rate at which a sample and hold function is updated. The morpher does not move continuously from one offset to another, but samples the coordinate in the morph space and holds it until the next sample is taken.

8 Routing, Patching, and Pitch Quantization

MOLEKULAR comes equipped with a wide variety of routing options. Use the [ROUTING](#) section to alter the signal chain and change the order in which the audio is being processed. MOLEKULAR gives you many options to be creative with the routing, with numerous different settings for parallel and serial routing as well as the ability to swap positions between the effect units.



The routing section

- (1) [Input](#): Turn to set the level of the input signal. This control is Snapshot independent and cannot be modulated.
- (2) [Dry](#): Turn to adjust the amount of dry signal to output. This control is Snapshot independent and cannot be modulated.
- (3) [FX](#): Turn to adjust the amount of wet signal to output. This control is Snapshot independent and cannot be modulated.

(4) **ROUTING**: Click the arrow icon next to the **ROUTING** label to open the routing selector at the center of the interface.

(5) **Pitch Quantization**: Click the tuning fork icon to open the **Pitch Quantization** page. For detailed information about pitch quantization, see section [18.4, Pitch Quantization](#).

(6) **Morph Quantization**: Click the morpher icon to open the **Morph Quantization** page. For detailed information about morph quantization, see section [17.4, Morph Quantization](#).

(7) **Flowchart**: This section changes to reflect the selected routing chart. Click and drag the numbers **1**, **2**, **3**, and **4** to swap their positions within the chart and thus the order in which their corresponding effect units are processed.

(8) **Latency**: Click to switch between displaying latency in values of samples or in milliseconds. The FFT based algorithms Spektral Hold, Spektral Shift, Spektral Smear, and Vokoder introduce latency that is internally compensated within MOLEKULAR. The overall latency is dependent on the used modules and routing. It is therefore recommended you keep an eye on the latency value, so you can compensate for it manually in your host software. For information about how to compensate for latency, refer to the documentation of your host software.

(9) **Scan**: Activate to introduce a crossfader to the routing chart. Click and drag the appearing crossfader to mix the DSP signals before the output. Use it creatively by modulating the position of the crossfader to add variation to the routing.

(10) **Filter**: Turn the color coded controls to introduce a filter to the corresponding DSP. Turn the control left of center position to add a low-pass filter. Turn the control right of center to add a high-pass filter. At center position no filtering is applied.

(11) **Delay**: Turn the color coded controls to introduce a delay to the corresponding DSP. Turn the control left of center to add delay and output the wet signal only. Turn the control right of center to add delay and to output the wet signal and the dry signal simultaneously.

(12) **Sync**: Activate to synchronize the delay to a note value set with the **Delay** control. When **Sync** is inactive, the value is set in milliseconds.

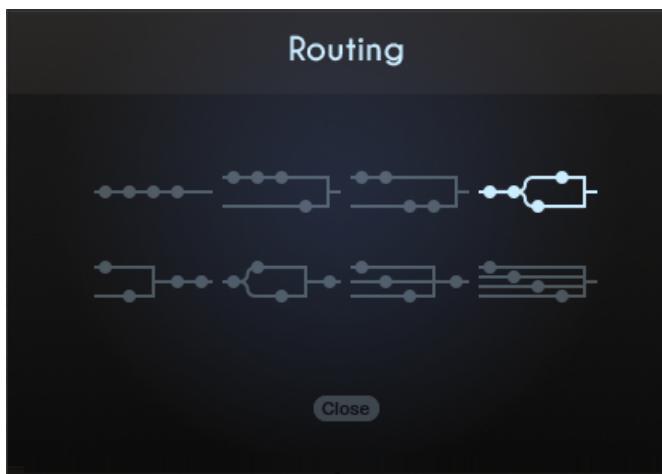
(13) **PATCH**: Activate to open the patch view. Use it to gain further control over the routing and, e.g., to add feedback. For detailed information, see section [18.3, Patching](#).

8.1 The Routing Selector

Use the routing selector to pick one of eight routing charts and change the order in which the effect units are processed. The output audio can vary drastically depending on the selected routing.



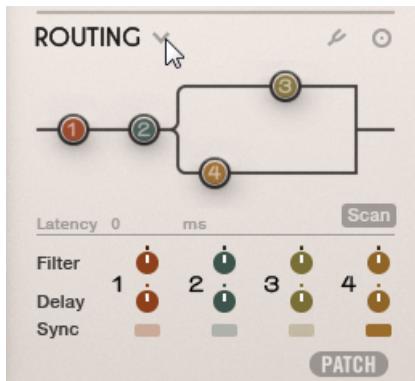
If you cannot hear the output, ensure that the Out control in each DSP footer is not set fully left, effectively blocking the signal chain.



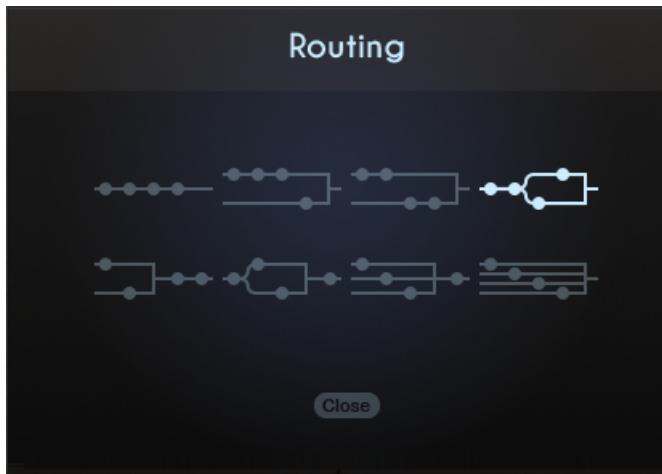
The routing selector, open at the center of the MOLEKULAR interface

To select a routing chart:

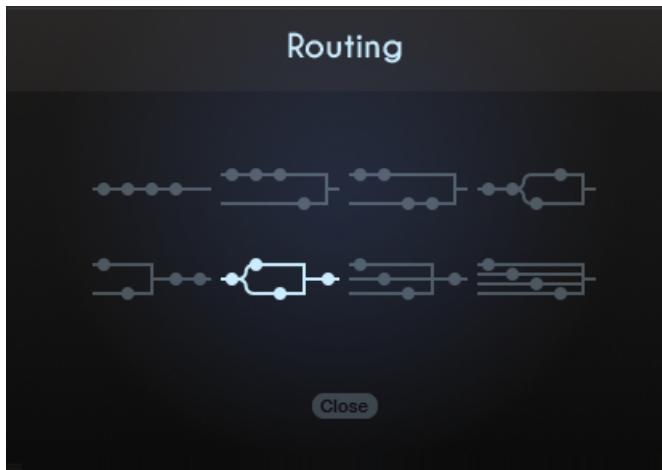
1. To open the [Routing](#) selector, click the down-pointing arrow next to the **ROUTING** label in the routing section.



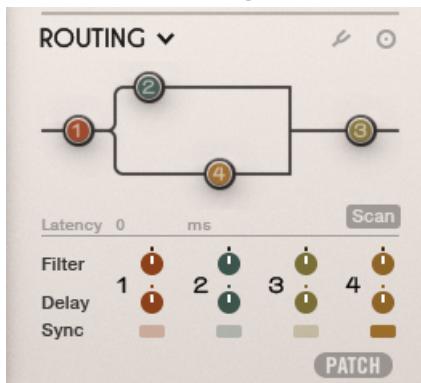
The [Routing](#) selector opens at the center of the interface.



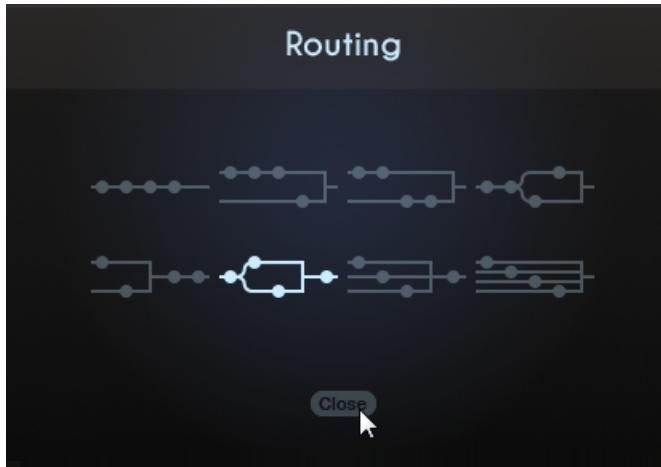
2. To load a routing chart, click the corresponding flowchart.



The selected routing chart is loaded in the **ROUTING** section.



3. To close the [Routing](#) selector, click [Close](#).



For more information about how to use routing, see sections [18.2, Using the Routing Section](#) and [18.3, Patching](#).

8.2 Using the Routing Section

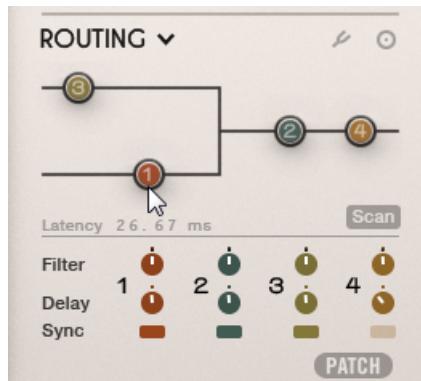
Beyond selecting one of eight routing presets, as described in section [18.1, The Routing Selector](#), you can also change the positions of the effect units within the flowchart and define the junction point at which the signals are mixed before being output. For information about the Filter and Delay controls, see section [18, Routing, Patching, and Pitch Quantization](#).

8.2.1 Swapping DSP Positions

- ▶ To swap the positions of the DSPs within the flowchart, click and drag the corresponding number to another number's position.



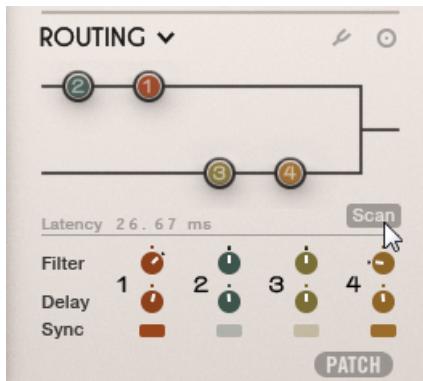
- The DSPs swap positions in the flowchart.



8.2.2 Adding a Crossfader

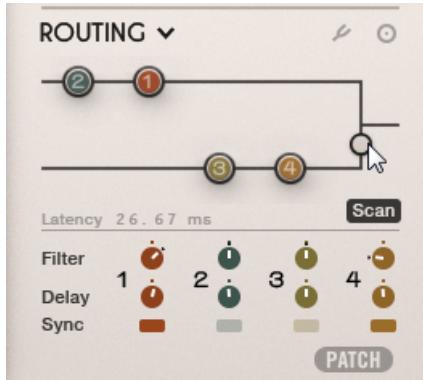
- ▶ To crossfade between the DSP signals and decide the mix that will be output:

1. To add a crossfader, click the **Scan** button.



The **Scan** button is not available for the single-channel serial routing.

2. To define the mix of the DSPs' output signals, click and drag the crossfader up or down.

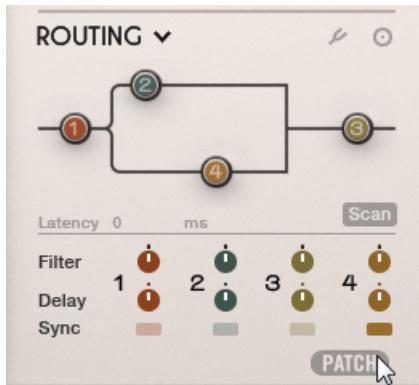


→ The output mix will now reflect the position of the crossfader.

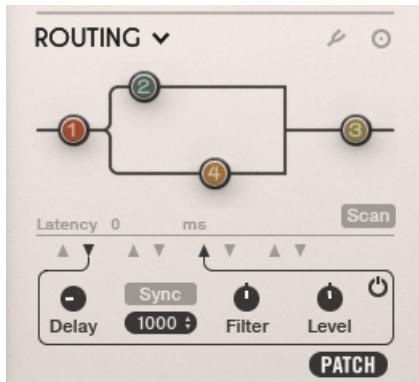
8.3 Patching

The patch view gives you further control over the routing by letting you patch together the input of one DSP with the output of another, just like a real patch cable on a modular system. Beyond this, you can also add yet another delay and multi-mode filter.

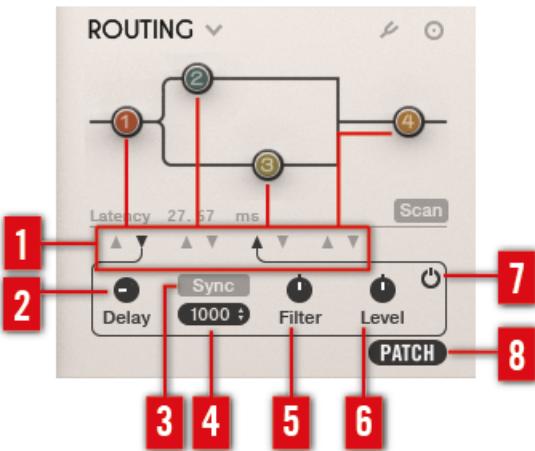
- To open the patch view, click the **PATCH** button in the bottom right corner.



- The patch view opens at the bottom of the **ROUTING** section.



Patch View



Patch view

(1) Patch points: Click these to select an input and an output to patch two effect units together. Each DSP has one corresponding input and one output. By patching two DSPs together you can create feedback or bypass one or more DSPs in the flowchart.

(2) Delay: Turn to add a delay effect to the routing chart. The delay will be introduced between the two DSPs connected via the patch cable.

(3) Sync: Activate to synchronize the delay to host tempo in note values. Define the note value with the delay time selector below the Sync button.

(4) Delay time: Click and drag to adjust the value of the Delay in either milliseconds or as a note value, depending of the setting of the Sync button.

(5) Filter: Turn to add filtering to the signal chain between the connected effect units. Turn it left of center position to apply a low-pass filter. Turn it right of center position to apply a high-pass filter. At center position no filtering is applied.

(6) Level: Turn to adjust the level of the patch settings, i.e. Delay and Filter. Leaving it at center position means Delay and Filter are not being applied. Turn left for phase inversion.

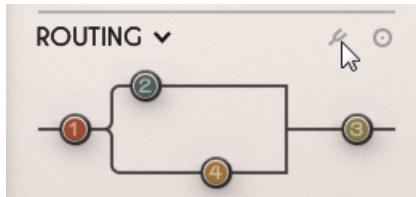
(7) On/off: Activate to switch patching on.

(8) **PATCH**: Click to switch between patch view and routing view.

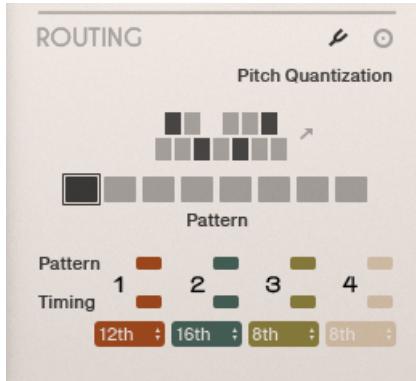
8.4 Pitch Quantization

One feature that really makes MOLEKULAR stand out is the ability to precisely quantize all pitch-related parameters of active DSPs to bespoke scales that you define using a pattern editor. For example, if you assign e.g. an LFO to modulate the pitch parameter of an effect unit (such as the one found on the Vocoder in DSP 1) while using pitch quantization, the DSP's output will be "drawn" toward the tones selected by you in the pattern editor, even though the LFO moves the **Pitch** control continuously through the frequency spectrum. Adding more keys to the pattern equals more quanta and thus a broader spectrum of tones. You can further decide the update rate of the quantization, by using the S&H function **Timing**. Use the **Pitch Quantization** page to gain access to controls for pattern selection.

- To open the **Pitch Quantization** page, click the tuning fork icon on the right side of the interface.



- The **Pitch Quantization** page opens in the **ROUTING** section.

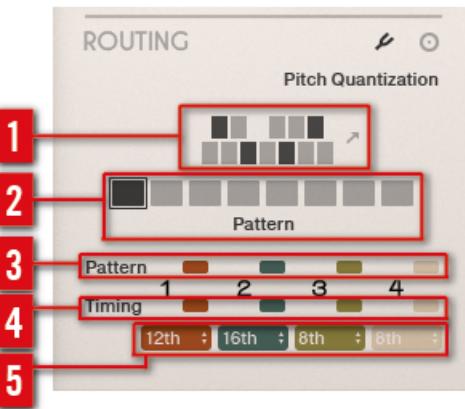




For information about **morph quantization**, see section [17.4, Morph Quantization](#).

The Pitch Quantization Page

The Pitch Quantization page gives you access to the following controls:



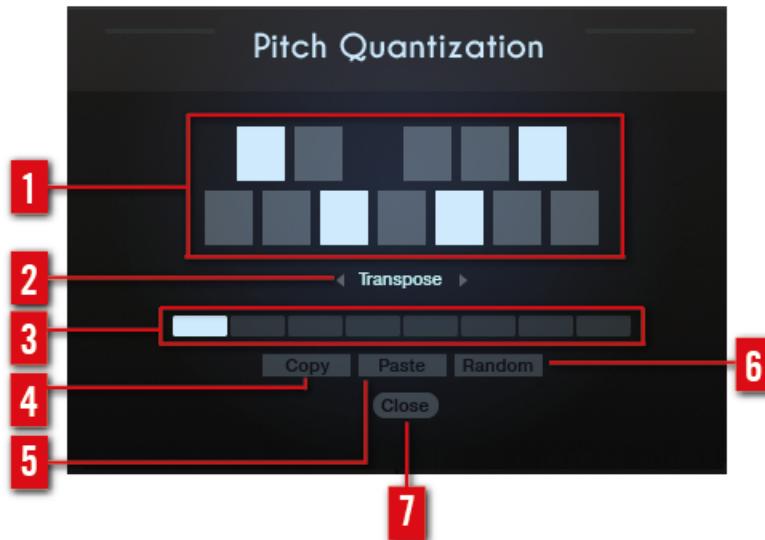
The Pitch Quantization page

- (1) **Pattern editor** (and **arrow icon**): Activate to open the pattern editor at the center of the interface. Use it to create custom scales according to which pitch-related parameters of selected DSPs will be quantized. For detailed information about the pattern editor, see the section below.
- (2) **Pattern selector**: Select one of eight patterns to play by clicking the corresponding button. A good tip is to modulate this feature using a step sequencer that switches between patterns.
- (3) **Pattern on/off** (1, 2, 3, and 4): Activate these color coded switches to use the selected pattern for the corresponding effect unit. When switched on, any pitch-related parameters of a selected DSP are quantized to the pattern you set with the pattern editor.
- (4) **Timing on/off** (1, 2, 3, and 4): Activate to sync the quantization pattern to the host tempo, using a note value set with the selector below.

(5) **Timing selector:** Click and drag to set a rate in note values by which a sample and hold signal is updated. The pitch is sampled and then held until the next sample is taken, at which point the pitch at its current value will be held until next sample is taken etc.

The Pitch Quantization Pattern Editor

When the [Pitch Quantization](#) pattern editor is open, you see two visual representations of the playing pattern: one on the [Pitch Quantization](#) page to the right and one being displayed at the center of the interface. Use the pattern editor at the center display to create and edit patterns, whereas the visual representation on the [Pitch Quantization](#) page is used to select what pattern to play.



Pitch Quantization pattern editor

(1) **Keys:** Click to activate the corresponding note in the octave. Pitch-related parameters will be quantized toward the selected keys.

(2) **Transpose:** Click the arrow-shaped buttons (left or right) to transpose the scale one step down or up along the displayed 'keyboard.'

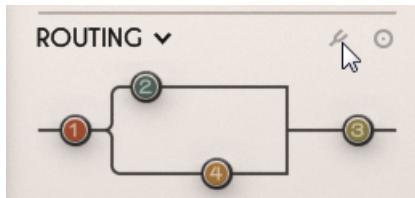
(3) **Patterns:** Click any of the eight buttons to access the values of the corresponding pattern.

- (4) [Copy](#): Click to copy the displayed values.
- (5) [Paste](#): Click to paste values placed in the clipboard.
- (6) [Random](#): Click to randomize all values.
- (7) [Close](#): Click to close the pattern editor.

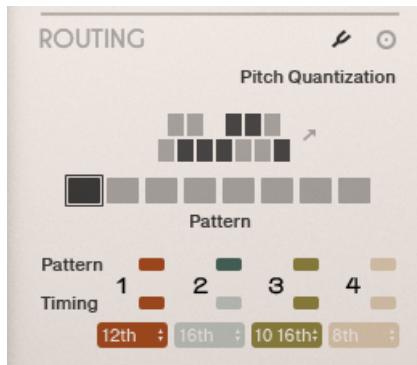
8.4.1 Applying Pitch Quantization

To apply pitch quantization:

1. To open the [Pitch Quantization](#) page, click the tuning fork icon on the right side of the interface.



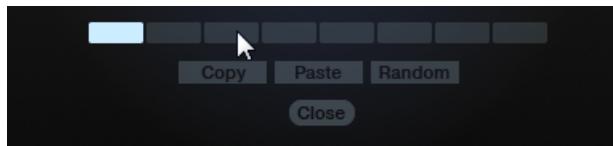
2. To open the [Pitch Quantization](#) pattern editor, click the visual representation of the pattern or the arrow icon next to it.



The pattern editor opens at the center of the interface.



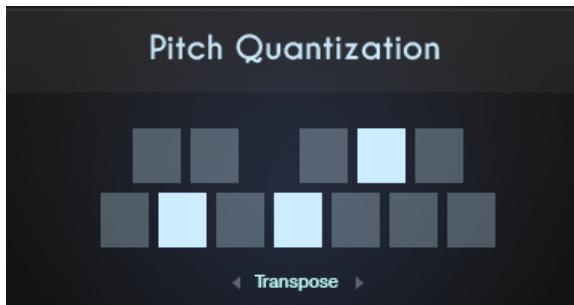
3. To select one of eight patterns for editing, click the corresponding button.



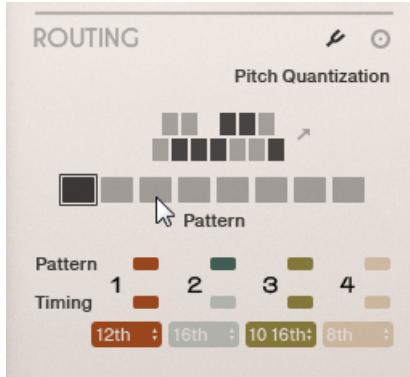
4. To set the tones to which pitch-related parameters of selected DSPs are quantized, click the virtual keys.



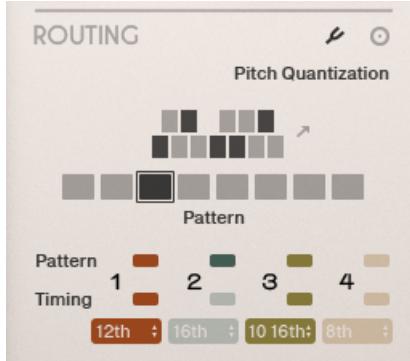
The turned-on keys illuminate.



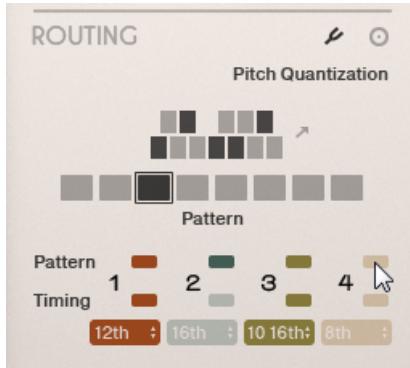
5. To quantize the tuning of DSPs' pitch-related parameters to the tones you selected, select the corresponding pattern on the [Pitch Quantization](#) page.



The [Pattern](#) changes to reflect the selection.



- To apply the quantization you just created to the DSPs of your choice, click their corresponding color coded **Pattern** switches on or off.

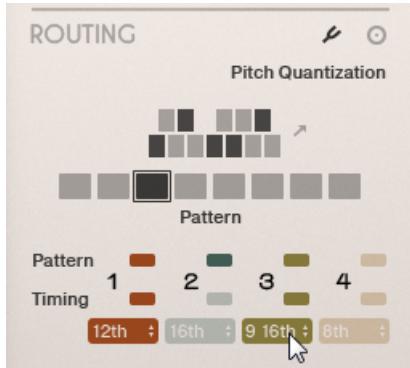


Ensure that the color coded **Pattern** switch is activated for the DSP you want to tune! If it is deactivated, the DSP's pitch-related parameters will not be quantized.

Applying Timing

The **Timing** control works similar to a sample and hold function, in that it lets you set a note value to decide how often the pitch will be updated.

- To set the rate at which the pitch is updated, activate the color coded **Timing** switch for the corresponding DSP and click and drag the selector below.



8.4.2 Modulating Pitch Quantization

One interesting way to use MOLEKULAR almost like an instrument is to create a set of different pitch quantization patterns and switching between these by modulating the pattern selector on the [Pitch Quantization](#) page.



Typical usage for modulating the pattern selector is to use a step sequencer as modulation source. For detailed information about the step sequencers and their parameters, see section [15.7.2, Step Sequencer](#).

- The [Pattern](#) selector now switches between the patterns—each containing the tones you selected—at the rate set by the modulator (but only updates the pitch at the rate set with [Timing](#)).



Ensure that the color coded [Pattern](#) switch is activated for the DSP you want to tune! If it is deactivated, the DSP's pitch-related parameters will not be quantized.

9 DSP Reference

In this section you find detailed information about the available effect units and their parameters.

9.1 Using the Effect Units



The four DSPs at the bottom half of the interface

At the bottom of the MOLEKULAR interface you will find four DSPs (digital sound processors) identifiable by color and number. Each of the DSPs contains up to 14 selectable effect units, with Dual Delay, Equalizer, Filter, Level, and Metaverb being available for each DSP. Beyond

those five recurring effect units, all four DSPs also contain their own unique effect units. All effect units have a header, a main section, and a footer. The header and footer both contain a number of parameters that are shared by all effect units (with only a few exceptions).



Press [Shift] on your computer keyboard while clicking and dragging a parameter to adjust the parameter in finer increments.

9.1.1 The Header

The controls located in the header of each DSP are identical to all effect units.



The header

(1) Effect title and DSP selector: The title of the loaded effect unit is always visible. Click to open the DSP selector at the center of the interface and load an effect unit into the corresponding DSP. Each of the four DSPs has its own individual list of available effect units.

(2) S (solo in place): Activate to solo the corresponding DSP. Doing so mutes the output of other DSPs. However, the output signal of DSPs located before the soloed DSP in the routing chart are sent through the soloed DSP. You can solo only one DSP at a time.

(3) On/off: Click to include or bypass the effect unit in the signal chain.

9.1.2 The Footer

The controls located in the footer of each DSP are identical to all effect units, unless stated otherwise in affected effect unit section further below.



The footer

(1) **Mute**: Activate to mute the corresponding DSP at the input. Switching **Mute** on and off via modulation can lead to interesting and unexpected effects! A few effect units have an **X.gate** button instead of a **Mute** button, for information about this feature; refer to affected units' sections further below.

(2) **Mix**: Turn to crossfade between the processed signal and the "dry" input signal. Set it fully left to hear only the "dry" input signal; set it fully right to hear only the processed signal. At center position you will hear both signals mixed 50/50. Use it together with **Wet** to achieve the sound you want.

(3) **Wet**: Turn to adjust the level of the processed signal. Use it together with **Mix** to get the sound you want.

(4) **Out**: Turn to set the output level of the corresponding effect unit. The signal will then be mixed with the outputs of the other DSP, depending on the selected routing. Note that the DSP's position in the routing chart also defines the behavior of the **Out** control, as turning it

fully left will stop the signal from all DSPs located before the affected DSP in serial routing. For detailed information on the routing system, see section [18, Routing, Patching, and Pitch Quantization](#).



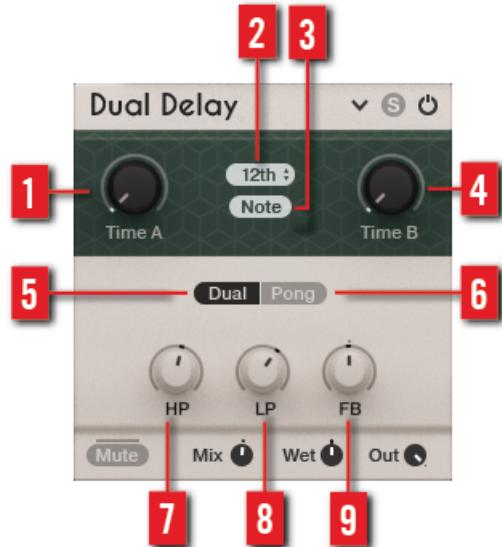
If you cannot hear the output, ensure that the Out control in each DSP footer is not set fully left, effectively blocking the signal chain.

9.2 Recurring Effect Units

The following five effect units can each be selected in all four DSPs, even simultaneously!

9.2.1 Dual Delay

A comprehensive delay effect that offers two delay units, A and B. These can be routed in two different ways: **Dual** and **Pong**. Feedback with hi- and low-pass filtering is provided, and delay times can be set in milliseconds or musical values.



Dual Delay

(1) **Time A**: Turn to set the delay time for Delay A, either in milliseconds or in multiples of the note value, depending on the settings of **Note** and **Note Value**.

(2) **Note Value**: If **Note** is selected, this selects the base delay time in musical values.

(3) **Note**: Activate to adjust the delay time in multiples of the value set with the **Note Value** selector. Deactivate it to set delay time in milliseconds.

(4) **Time B**: Turn to set the delay time for Delay B, either in milliseconds or in multiples of the note value, depending on the settings of **Note** and **Note Value**.

(5) **Dual**: Select this to feed the left input channel to Delay A and the right input channel to Delay B, resulting in independent delay times for the left and right input channels.

(6) **Pong**: Select this to feed the input to Delay A. The delay is fed both to the output as well as to the input of Delay B. The output of A is panned to the left channel and B is panned to the right channel.

(7) **HP**: Turn to adjust the hi-pass filter's cutoff frequency. The filter is placed in the feedback path.

(8) **LP**: Turn to adjust the low-pass filter's cutoff frequency. The filter is placed in the feedback path.

(9) **FB**: Turn to adjust the feedback amount. Feedback routing is context sensitive and adapts to whether you have selected **Dual** or **Pong** as the delay mode.

9.2.2 Equalizer

An equalizer that allows you to smoothly crossfade from low shelf via peak/notch to high shelf.



Equalizer

- (1) **Gain**: Turn to adjust boost/cut by +/- 18dB.
- (2) **Type**: Turn to crossfade from low shelf via peak/notch to high shelf.
- (3) **Freq**: Turn to set the EQ frequency in pitch values. Turn left for lower pitch and right for higher.
- (4) **Q**: Turn to adjust the bandwidth of the peak/notch EQ.

9.2.3 Filter

A multi-mode filter that lets you select and crossfade between low-pass, bandpass, and high-pass filtering.

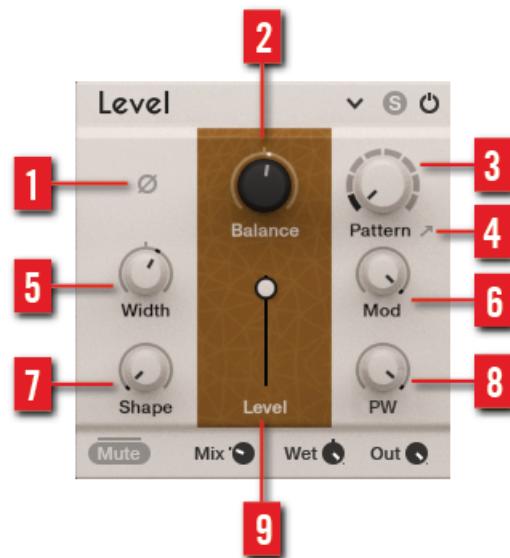


Filter

- (1) **Cutoff**: Turn to set the cutoff frequency.
- (2) **Resonance**: Turn to adjust the amount of resonance.
- (3) **Filter Type**: Turn to set the filter type, from low-pass filter (fully left) via bandpass filter at center position to high-pass filter (fully right).

9.2.4 Level

A level-control tool with stereo-width adjustment and sequencers for level, pan and pulse width. Pulse width allows you to create "gating" effects.

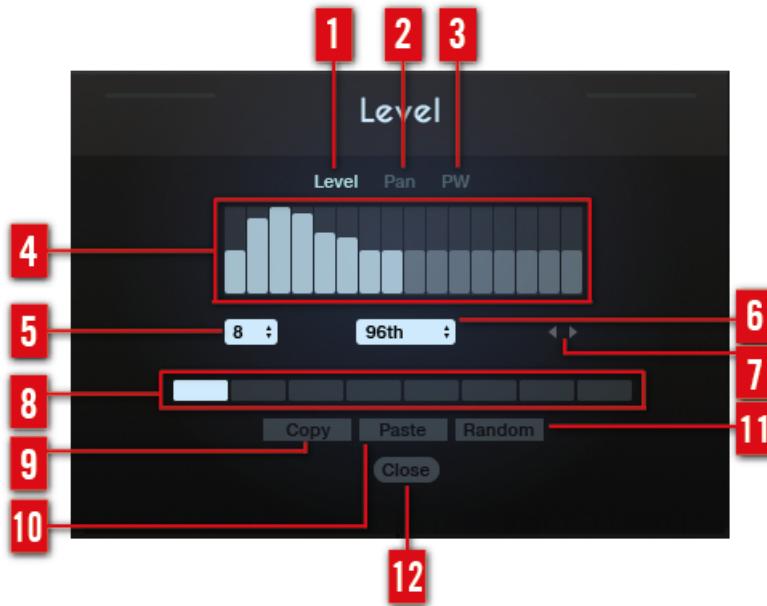


Level

- (1) **∅** (Phase inversion): Activate to invert the phase of the output.
- (2) **Balance**: Turn in either direction to adjust panning.
- (3) **Pattern**: Turn to select one of eight corresponding patterns. Set the patterns using the pattern editor.
- (4) **Arrow icon**: Click to open the pattern editor at the center of the interface. For detailed information about the pattern editor, see the section below.
- (5) **Width**: Turn to adjust stereo width using an M/S encoder/decoder. Turn fully left for mono (mid only) and fully right for wide (side only).
- (6) **Mod**: Turn to adjust the amount by which the level and pan sequencers control the values.
- (7) **Shape**: Turn to adjust fader and modulation response from linear (fully left) to exponential (fully right), and to add slew-rate limiting (smoothing).
- (8) **PW**: Turn to set the base value for pulse width; the value set in the pattern editor is then added to the base value. Create stuttering effects by using **PW** to control the gate length of patterns created in the Level pattern editor.

(9) [Level](#): Slide to adjust the output level.

Level Pattern Editor



Level pattern editor

- (1) [Level](#): Click and drag the sliders to set the level stepwise.
- (2) [Pans](#): Click and drag the bipolar sliders in either direction to adjust the panning stepwise.
- (3) [PW](#): Click and drag the sliders to set values for pulse width stepwise.
- (4) **Steps**: Click and drag the sliders to set values for the selected steps ([Level](#), [Pan](#), or [PW](#)).
- (5) **Step amount**: Click and drag to set the amount of steps from 1 to 16.
- (6) **Step rate**: Click and drag to set the rate in note values at which the sequence plays.
- (7) **Arrows**: Click to move the entire sequence one step in either direction.
- (8) **Patterns**: Click either button to access the values of the corresponding pattern.

- (9) [Copy](#): Click to copy the displayed values.
- (10) [Paste](#): Click to paste values placed in the clipboard.
- (11) [Random](#): Click to randomize all values.
- (12) [Close](#): Click to close the pattern editor.

9.2.5 Metaverb

Native Instruments MASCHINE's well-known Metaverb is a reverb effect with a slightly synthetic sound that gives you separate EQ controls for low and high frequencies.



Metaverb

- (1) [Size](#): Turn to adjust the size of the virtual space.
- (2) [Pan](#): Turn either direction to pan the dry signal. This is useful because the dry signal cannot be panned after the effect without panning the reverb itself, which can sound unnatural.
- (3) [Low](#): Turn to adjust the level of low frequencies in the reverb tail.
- (4) [High](#): Turn to adjust the level of high frequencies in the reverb tail.

9.3 DSP 1 Effect Units

The leftmost effect slot is dedicated to spectral effects. Use these effects to control and alter the spectral data of the input signal.

9.3.1 Dual Comb

Two comb filters, A and B, are arranged so that you can crossfade from parallel to serial routing. Comb filter A has its center frequency set absolutely in MIDI Note Numbers, and comb filter B's center frequency is set as an interval relative to comb filter A's value in a range of -36 to +36 semitones. Both comb filters allow manipulating the odd/even harmonic balance.



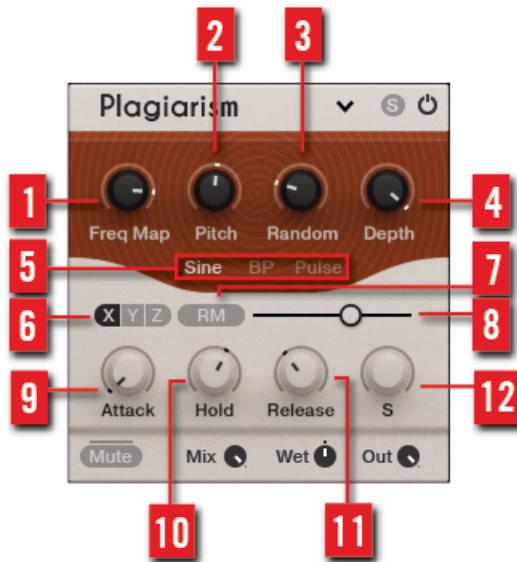
Dual Comb

(1) **Mix:** Turn to crossfade between the outputs of comb filters A and B. In parallel mode this means you're crossfading between two separate comb filters. In serial mode, this means you're crossfading between comb filter A and the result of feeding comb filter A through comb filter B.

- (2) **Pitch**: Turn to set the pitch of comb filter A in semitones.
- (3) **Parallel/Serial**: Click to select either parallel or serial routing for the two comb filters.
- (4) **Reso (A)**: Turn to set the resonance amount for comb filter A.
- (5) **Ø (A/B inversion)**: Activate to invert the value of **Odd/Even** for comb filter B.
- (6) **Interval A**: Turn to set the pitch for comb filter B as an offset of the range -36 to +36 semitones relative to comb-filter A.
- (7) **Reso (B)**: Turn to set the resonance amount for comb filter B.
- (8) **Odd/Even**: Turn to adjust the odd/even harmonics balance for both comb filters simultaneously.

9.3.2 Plagiarism

The input signal's amplitude is measured using an envelope follower. The measured RMS data is used to trigger 16 envelopes, which in turn are used to play back 16 “voices.” The voices are commonly switched between bandpass, sine-wave, or pulse-wave modes. The pitch or center frequency of each voice is generated from a base pitch plus per-voice offset that is read from a look-up table with various interval sequences. For example, the 16 voices can be configured to play the base pitch and its first 15 harmonics as sine waves. In pulse-wave mode, the envelope is additionally used to modulate pulse width. A ring modulator and a frequency randomization function are available as well.



Plagiarism

(1) **Freq Map**: Turn to smoothly switch between sets of values that are used as center frequencies or pitches for the 16 voices. Available sets are natural harmonics, reversed order harmonics, odd/even harmonics, stacked fifths, exponential, minor, major, major 7, minor 7, detuned octaves, and stacked 2nds.

(2) **Pitch**: Turn to shift the base pitch/frequency of all voices simultaneously up or down.

(3) **Random**: Turn to add small random deviations to the voice pitches/frequencies, resulting in a more lush sound.

(4) **Depth**: Turn to scale the envelope follower output before analysis. When turned left fewer voices are triggered, when set to center position all 16 voices are triggered, and turning the control fully right tends to (depending on the combined settings of a number of parameters) wrap the voice allocation around, resulting in strongly moving spectra.

(5) **Sine/BP/Pulse**: Activate one of these to select whether bandpass filters, sine waves or pulse waves are used to synthesize the effect signal.

(6) **RM** (ring modulation): Click to switch the ring modulator on or off.

(7) **RM Pitch:** Slide this in either direction to tune the ring modulator up or down.

(8) **XIYZ:** Click either letter to select one of three analysis/triggering modes.

- In **X** mode, the envelope follower output is compared to 16 linearly spaced steps, and all steps with a threshold lower than the envelope follower output are triggered (kind of like how LEDs light up in a level meter). This mode allows triggering of all voices simultaneously.
- **Y** mode triggers only one voice at a time (though they will typically overlap while their envelopes are running) based on the level of the envelope follower signal (imagine only the highest level LED in a meter lighting up), and has a shape (**S**) parameter associated that smoothes the envelope follower signal before analysis. This results in “arpeggios” across voices.
- **Z** mode triggers one voice every millisecond. Which voice is triggered is determined by sampling the envelope follower signal at the same time the trigger is generated. Technically, this triggers only one voice at a time, but in practical terms it sounds like multiple voices triggered simultaneously.

(9) **Attack:** Turn to adjust the attack time on all voices' amplitude envelopes simultaneously.

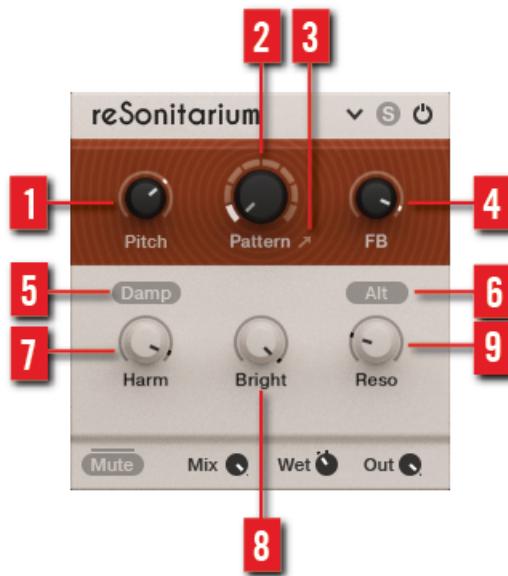
(10) **Hold:** Turn to adjust the hold time on all voices' amplitude envelopes simultaneously.

(11) **Release:** Turn to adjust the release time on all voices' amplitude envelopes simultaneously.

(12) **S:** Turn to adjust the smoothing shape for **Y** mode. Note that you have to first activate **Y** mode in the **XIYZ** selector to turn the **S** control.

9.3.3 reSonitarium

The input signal is passed through four parallel feedback-delay-based resonators, each of which is passed through its own resonant low-pass filter and delay. The resonators allow blending between odd and even harmonics, realized by two parallel delay lines an octave apart, which are subtracted from each other and crossfaded between. The per-resonator low-pass filter's cutoff frequency is set relative to the resonator pitch. Resonator pitches are set using a base pitch and three intervals. You can set intervals, delay times and per-resonator panning and store and recall several patterns using a sequencer. Using the **Damp** parameter, you can set resonator levels to be scaled in a 1/f fashion, and odd/even harmonic balance value can be inverted for odd resonator numbers.



reSonitarium

(1) **Pitch**: Turn to set the pitch of resonator #1, and thus the base pitch of the effect. The intervals in the pattern editor are added to this value to generate the pitches for the three other resonators. (2) **Pattern**: Turn to select one of eight patterns. Patterns include values for intervals, delays and pans of the resonators.

(3) **Arrow icon**: Activate the arrow to open the pattern editor at the center of the interface. For detailed information about the pattern editor, see below.

(4) **FB**: Turn to adjust the amount of resonator feedback.

(5) **Damp**: Activate to set the output level of resonators to be reduced relative to their pitch, imitating the way natural signals are structured. This way a resonator that is twice the frequency of the base pitch will play back at half the level of the base pitch etc.

(6) **Alt**: When this is switched on, the **Harm** parameter is inverted for resonators 2 and 4.

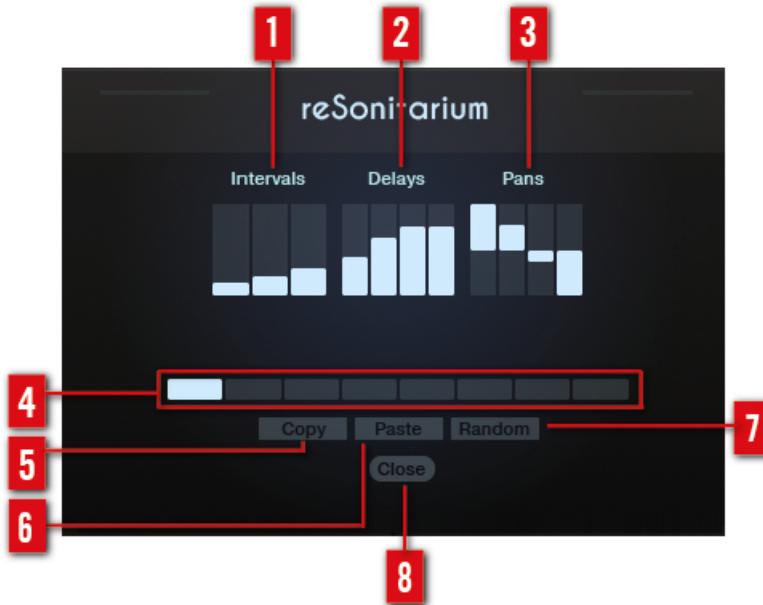
(7) **Harm**: Turn to adjust the balance between even and odd for the harmonics created in the resonator.

(8) **Bright**: Turn to set the cutoff frequency of the low-pass filters relative to the pitch of the resonators. Set fully left, the cutoff frequency is at the same frequency as the resonator. Turning the control clockwise tunes it upwards.

(9) **Reso**: Turn to adjust the resonance amount of the low-pass filters.

reSonitarium Pattern Editor

Use the pattern editor to display and adjust the settings for each of the eight patterns. Parameters are available for resonator harmonic intervals, delay times, and pans.



reSonitarium pattern editor

(1) **Intervals**: Click and drag the sliders in the window to adjust the intervals of the three resonator voices relative to the base pitch set using the [Pitch](#) control.

(2) **Delays**: Click and drag the sliders in the window to set the delay times for each of the four delays individually.

- (3) **Pans:** Click and drag the bipolar sliders in either direction, up or down, to pan the corresponding delay left or right.
- (4) **Patterns:** Click either button to access the values of the corresponding pattern.
- (5) **Copy:** Click to copy the displayed values.
- (6) **Paste:** Click to paste values placed in the clipboard.
- (7) **Random:** Click to randomize the values of each tap for the selected set.
- (8) **Close:** Click to close the pattern editor.

9.3.4 Spektral Hold

The signal is passed through a 256 or 512 band FFT (fast Fourier transform) and the resulting spectral data is written into a buffer. When a gate signal is received, the buffer content is frozen and played back in a loop (where the amplitude values are played back as is and the phase values are used to generate a running phase), resulting in the instantaneous spectrum being sustained for as long as the gate signal is held. Alternately, you can switch to "continuous mode," where a trigger will cause the buffer to be frozen until the next trigger is received. The buffer can be adjusted in length from one to 16 FFT frames, and the phase information can be randomized for a less static sound. Finally, an iFFT (inverse fast Fourier transform).



The FFT introduces a bit of latency, which is displayed in the [ROUTING](#) section. For information on how to compensate for latency in the host, refer to the documentation of your host software.



Spektral Hold

(1) **Frames**: Turn to adjust how many spectral frames are “frozen” when [Play](#) is on, similar to the “loop length” in time-based freezers.

(2) [Play](#): Click (or trigger via modulation) to activate the spectral hold.

(3) [Random](#): Turn to add random phase modulation to make the sound less static when Hold is on.

(5) [∞](#): When this is on, the [Play](#) control is interpreted as a trigger. The spectrum is sampled when [Play](#) goes high, and is held until the next time [Play](#) goes high. When this is off, [Play](#) is interpreted as a gate signal, and the spectrum is held only while [Play](#) is high (on). With [Play](#) low (off), the input signal is played back unaltered.

(4) [FFT size](#): Click the switch to select between a 256 and a 512 band FFT, influencing both the sound as well as the length of the frames.

(6) [X.gate](#): Use this feature to define the behavior of the effect unit when the [Play](#) button is inactive. Activate [X.gate](#) to pass the input through to the output while [Play](#) is inactive. Deactivate [X.gate](#) to mute the output, unless the [Play](#) button is active.

9.3.5 Spektral Shift

The input is passed through a 256 or 512 band FFT (fast Fourier transform) and the resulting spectral data is written into a buffer. The spectral data is shifted up or down, by means of reading from the buffer with an offset in samples (which corresponds to a bin-shift) and muting all invalid (wrapped-around) bins. Additionally, an option is provided to allow only bin $m \cdot N$ to be heard; a sample counter counts from 0 to N , and when N is reached, the currently playing bin is passed and all other are muted. If N is not an even number, this results in the active bins “moving” through the spectrum; even values for N result in stationary results. Finally, an iFFT (inverse fast Fourier transform) is performed.



The FFT introduces a bit of latency, which is displayed in the [ROUTING](#) section. For information on how to compensate for latency in the host, refer to the documentation of your host software.



Spektral Shift

(1) **N**: Turn to add a “spectral arpeggio” for values greater than one. Essentially, only every Nth spectral band is allowed to play. Even values create static textures, odd values create spectral arpeggios.

(2) **Shift L**: Turn to set the amount of spectral shifting for the left output channel.

(3) **Shift R**: Turn to set the amount of spectral shifting for the right output channel.

(4) **FFT size**: Click to switch between a 256 and a 512 band FFT. This setting influences the resolution of the **Shift** parameters and the **N** parameter.

(5) **Link**: Activate to override the right-channel setting and apply the same shift amount to both channels.

9.3.6 Spektral Smear

The signal is passed through a 256 or 512 band FFT (fast Fourier transform), the resulting spectral data is split into two parallel streams, based on a threshold that the amplitude values are compared against. The two streams are called **Above** and **Below**, and each has options to average its contents over time, using **Attack** and **Release** controls. Phase data can be randomized and all data below or above a lower and upper cutoff point set in “bins” can be discarded. Finally, crossfading between **Above** and **Below** is applied, and an iFFT (inverse fast Fourier transform) is performed.



The FFT introduces a bit of latency, which is displayed in the **ROUTING** section. For information on how to compensate for latency in the host, refer to the documentation of your host software.



Spektral Smear

- (1) **Threshold:** Turn to set the amplitude threshold according to which the spectral bands are separated into [Above](#) and [Below](#).
- (2) **LC** (low cutoff): Slide to discard spectral bands below the set frequency.
- (3) **HC** (high cutoff): Slide to discard spectral bands above the set frequency.
- (4) **Random:** Turn to add random phase modulation to create a more diffuse sound.
- (5) **FFT size:** Click to switch between a 256 and a 512 band FFT. A broadband FFT allows more accurate detection of the Above/Below state, but instead gives you lower time resolution.
- (6) **Above/Below:** Crossfade from [Above](#) through the original mix at center position and finally to [Below](#).
- (7) **Attack (Above):** Turn to control the amplitude averaging attack time for [Above](#).
- (8) **Release (Above):** Turn to control the amplitude averaging release time for [Above](#). Long values tend to create a “spectral reverb” type sound.
- (9) **Attack (Below):** Turn to control the amplitude averaging attack time for [Below](#).

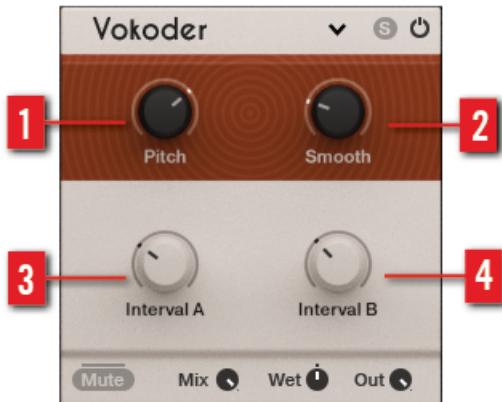
(10) **Release (Below)**: Turn to control the amplitude averaging release time for **Below**. Long values tend to create a “spectral reverb” type sound.

9.3.7 Vocoder

The FFT (fast Fourier transform) representations of the input signal and of three sawtooth oscillators are processed to achieve a vocoder effect. The processing is weighted towards the high frequency in order to reduce the effect of 1/f slope multiplication. For a more synthetic effect, the phase data of the input signal is discarded and only that of the oscillators is used. A control for smoothing the amplitude envelopes of the spectral bands is available.



The FFT introduces a bit of latency, which is displayed in the **ROUTING** section. For information on how to compensate for latency in the host, refer to the documentation of your host software.



Vocoder

- (1) **Pitch**: Turn to set the pitch in note names for oscillator 1.
- (2) **Smooth**: Turn to apply smoothing to the amplitudes of the spectral bands. Set low to track the input signal's features more accurately and high for more drone/pad-like results.
- (3) **Interval A**: Turn to set the pitch for oscillator 2 as an interval relative to oscillator 1.
- (4) **Interval B**: Turn to set the pitch for oscillator 3 as an interval relative to oscillator 2.

9.4 DSP 2 Effect Units

The DSP located second from left contains mostly effects that allow you to modify temporal structures. Use these modules to play with time and timing or to add interesting space to your mix.

9.4.1 Angel Delay

A delay effect with an adjustable number of taps (or tape heads), from three to 15, enabling delay in a number of different rhythms. Delay times for the taps are calculated from two macro functions called [Time](#) and [Shape](#). Generally, the taps are calculated from TapNumber multiplied by [Time](#). With [Shape](#) set to 1, the taps are linearly spaced [Time](#) milliseconds apart. [Shape](#) can be used to change the spacing from linear to exponentially longer or exponentially shorter with rising tap number. The output levels of the taps can be shaped linearly or exponentially rising or falling, or constant, or in between values. Means are provided to set panning adjustably dependent on tap number, including linear and alternating modes.



Angel Delay

(1) **Time**: Turn to set the base delay time for the effect. If **Shape** is set to center position, all taps are <Time> milliseconds apart. I.e., if you set Time to, e.g., 10ms, tap #1 will have a delay time 10ms, tap #2 20ms, tap #3 30ms etc.

(2) **Taps**: Turn to set the number of taps. Set fully left, three taps are used; set fully right, 15 taps are used.

(3) **Shape**: Turn to adjust the delay times of the taps relative to the base time set with **Time**. At center position, all taps are equally spaced (set spacing with the **Time** control). Turn the control left to make the delay time progressively longer for rising tap number. Turn the control right to make the time between taps progressively shorter (kind of like a “bouncing ball delay”).

(4) **Exp**: Activate to use an exponential level curve for the shape of the **Amp** parameter. Deactivate it to set the levels to linear stepping.

(5) **L/R/Alt 1/Alt 2/Alt 3/Alt 4 (panning selector)**: Click either radio button to select the corresponding panning mode for the taps:

- **L/R**: Select this to pan the first tap hard left, the last tap hard right, and to place the remaining taps linearly between left and right.
- **Alt 1**: Select this to pan odd-numbered taps to the left and even-numbered taps to the right.
- **Alt 2**, **Alt 3**, and **Alt 4** are basically the same, with the difference that groups of two, three or four taps are panned left/right. E.g., in **Alt 3**, the first three taps are panned left, the second group of three is panned right, the third group of three is panned left etc.

(6) **Amp**: Turn to adjust the levels of the taps. In center position, all taps have the same level. Turn the control left to cause the levels to be ramped up / faded in, so the first tap will be low in volume and the following taps get progressively louder. Turn the control right to reverse this behavior, where the first tap is at full level and successive taps are progressively lower in level (think fade-out).

(7) **Pan**: Turn to control the amount of panning applied to the taps, depending on the state of the **L/R/Alt 1/Alt 2/Alt 3/Alt 4** radio buttons. In center position, no panning is applied. Turn it fully right to pan taps hard left and hard right, and set the control to negative values to reverse the **L/R** mapping.

9.4.2 Band Delays

The input signal is passed through a stereo eight-tap delay, each tap being passed through its own bandpass filter, panning and amplitude scaling section. A pattern editor is provided, where you can adjust delay times, filter frequencies, panning and amplitude values. These four sets of values each consist of eight parameter values. The combined settings of these values sets together make up one pattern. 16 patterns may be stored and selected automatically via modulation. Delay times may be set in milliseconds or rhythmic values, and the maximum delay time may be set globally for all eight “voices” (each corresponding to one tap), as multiplier for the individual delay time values. Filter frequencies are set in MIDI note number, and the eight values may be shifted up or down globally using a [Transpose](#) control.



Band Delays

- (1) **Transpose:** Turn to shift all filters up or down by -24/+24 semitones.
- (2) **Pattern:** Turn to select one of eight patterns. Each pattern contains amp levels, panning, delay time, and cutoff frequency data for all eight taps simultaneously.
- (3) **Arrow icon:** Click to open the pattern editor at the center of the interface. For detailed information on the pattern editor, see section below.
- (4) **Reso:** Turn to adjust the amount of resonance for all eight bandpass filters simultaneously.

(5) **Note**: Activate to adjust delay times as multiples of a note value (which is set with the note value selector below). Deactivate it to adjust delay time in milliseconds.

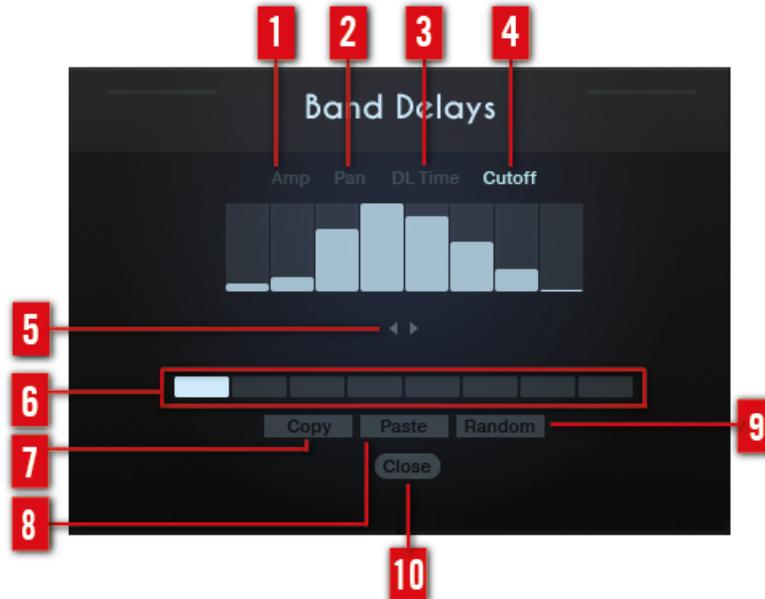
(6) **Note value**: With **Note** activated, click and drag to select the note value used as basis for the delay time. With **Note** deactivated, click and drag to select delay time in milliseconds.

(7) **Delay Sum**: Turn to set the maximum delay time for all taps (the delay times set in the pattern editor are scaled accordingly).

(8) **Pan Mod**: Turn to scale the panning amount set in the pattern editor. In center position, no panning is applied and all taps are centered. Set fully right, the panning data stored in the pattern is applied. Set fully left, the pattern panning data is applied but inverted (left becomes right and vice versa).

Band Delays Pattern Editor

Use the pattern editor to set values for amp levels, panning, delay times and cutoff frequency. Eight patterns are available and you can copy and paste values between them.



Band Delays pattern editor

(1) **Amp**: Select this set to display and adjust amp levels. Click and drag the sliders in the center window to set values for each of the eight taps individually.

(2) **Pan**: Select this set to display and adjust panning. Click and drag the bipolar sliders in the center window in either direction to pan the eight taps individually.

(3) **DL Time** (delay time): Select this set to display and adjust delay times. Click and drag the sliders in the center window to set values for each of the eight taps individually.

(4) **Cutoff**: Select this set to display and adjust the cutoff frequencies of each bandpass filter. Click and drag the sliders in the center window to set values for each of the eight taps individually.

(5) **Arrow buttons**: Use these to move all values of the selected set one step in either direction.

(6) **Patterns**: Click either button to access the values of the corresponding pattern.

(7) **Copy**: Click to copy the displayed values.

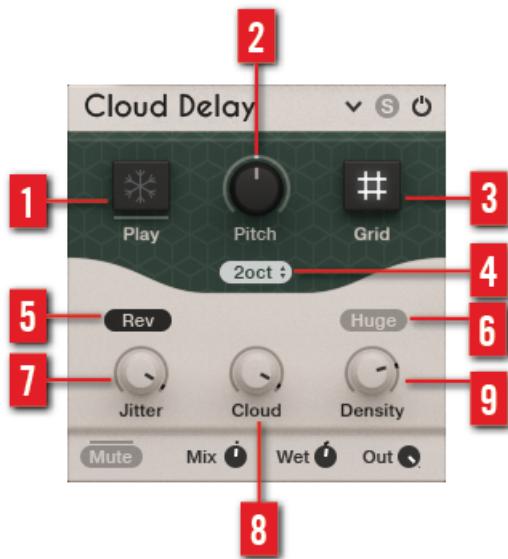
(8) **Paste**: Click to paste values placed in the clipboard.

(9) **Random**: Click to randomize the values of each tap for the selected set.

(10) **Close**: Click to close the pattern editor.

9.4.3 Cloud Delay

A granular cloud delay effect with macro control of the parameters of the REAKTOR Grain Cloud Delay.



Cloud Delay

(1) [Play](#): Click (or trigger via modulation) to freeze the grain delay buffer. In doing so you select the audio content that will be used as source material for the grains.

(2) [Pitch](#): Turn to transpose the grains.

(3) [Grid](#): Activate to quantize [Pitch](#) in semitones, rather than cents. [Grid](#) allows you to easier achieve clean-sounding intervals.

(4) [Pitch range](#): Click and drag to adjust the range of the [Pitch](#) control. Possible values are one semitone, one octave, two octaves, and four octaves.

(5) [Rev](#): Activate to play back grains in reverse.

(6) [Huge](#): Activate to elongate the range of delay time and grain length of the [Cloud](#) control.

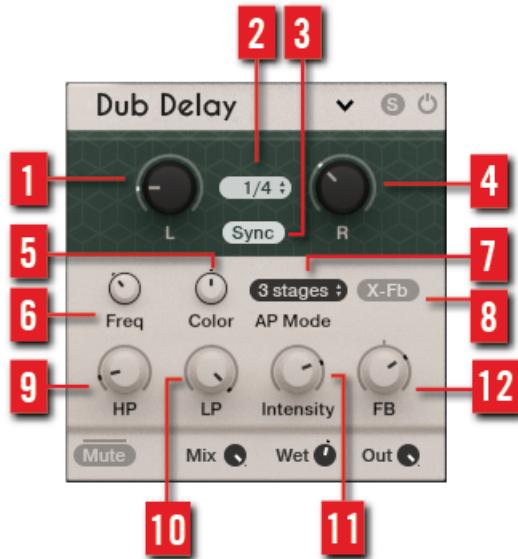
(7) [Jitter](#): Turn to adjust pitch, time, length, distance, and panning jitter simultaneously. Low values give a static sound; high values give a more diffused sound. This control interacts with [Cloud](#) and [Density](#) to ensure that the output is useful.

(8) [Cloud](#): Turn to adjust grain length and delay time. The control interacts with [Jitter](#) and [Density](#). Click the [Huge](#) button to adjust the range of this control.

(9) **Density**: Turn to adjust the distance between grains, and thus the number of grains audible simultaneously.

9.4.4 Dub Delay

A stereo delay with independent control of the left and right channel delay times, feedback and cross-feedback paths. Beyond this you can also apply heavy processing to the feedback path(s). Both high-pass and low-pass filters are applied to the feedback path, where the low-pass filter features resonance. Also, an oversampled saturation circuit as well as a three-stage all-pass filter with resonance and selectable number of stages is placed in the feedback path. A macro function called **Intensity** allows you to adjust saturation threshold, filter ripple/nonlinearity and filter resonance simultaneously.



Dub Delay

(1) **L**: Turn to adjust the delay time for the left channel. Delay time is adjusted in milliseconds or multiples of a note value, depending on the state of the **Sync** switch.

(2) **Note selector**: If **Sync** is activated, Click and drag to select the note value used for adjusting delay time. If **Sync** is deactivated, Click and drag to adjust delay time in milliseconds.

(3) **Sync**: Activate to adjust delay time in multiples of the note value with the Note selector. Deactivate it to adjust delay time in milliseconds.

(4) **R**: Turn to adjust the delay time for the right channel. Delay time is adjusted in milliseconds or multiples of a note value, depending on the state of the **Sync** switch.

(5) **Color**: Turn to adjust the resonance amount of the all-pass filter.

(6) **Freq**: Turn to adjust the center frequency of an all-pass filter in the feedback path.

(7) **AP Mode**: Click and drag to select one-, two- or three-stage all-pass filtering. The more stages you let the signal pass through, the more notches will occur in the audio and the sonic character will change.

(8) **X-Fb**: Activate to set feedback type to cross-feedback. I.e., the left channel is fed back into the right channel, and vice versa.

(9) **HP**: Turn to adjust the cutoff frequency of the feedback path's high-pass filter.

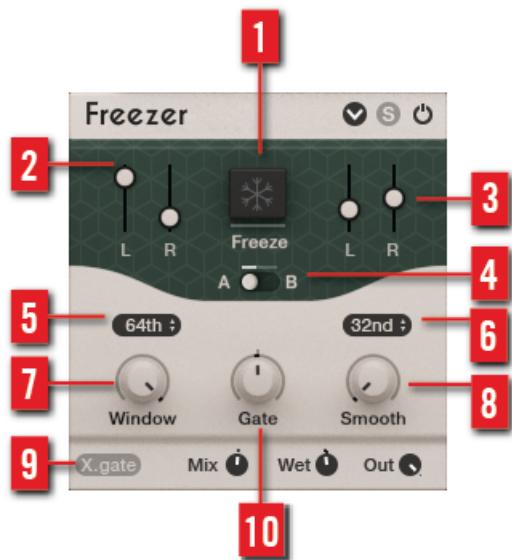
(10) **LP**: Turn to adjust the cutoff frequency of the feedback path's low-pass filter.

(11) **Intensity**: Turn to adjust several parameters "under the hood"—e.g. low-pass filter resonance and saturation—simultaneously. Set low for a clean sound, medium for a warm tone, and high for screaming tape-delay-esque feedback textures.

(12) **FB**: Turn to adjust the feedback amount.

9.4.5 Freezer

Whenever a gate signal goes high (triggers on), the input signal is sampled for an adjustable rhythmic duration, and looped until the gate signal goes low (off). Duration can be set independently for the left and right channels, and you can set and switch between two sets of L/R values, A and B. The playback loop can have fades applied to its start and end, a process which is commonly referred to as "windowing," and the **A/B** switch smoothed to prevent switching artifacts. Additionally, you can choose whether to mute the output when no gate is present, or whether to pass the input through when gate is turned off. Finally, a "gater" function is available, which allows to mute playback in one of two modes. In the first mode, the audio is turned on at the beginning of the playing loop and turned off after an adjustable percentage of the loop has been played. In the second mode, the same principle applies but loop lengths of the other set of values are used, so A will be gated using the values of B and vice versa.



Freezer

- (1) **Freeze**: Activate to sample and loop the input signal for an adjustable duration of time.
- (2) **L and R (A)**: Click and drag to adjust the length of the audio section to be looped in looper **A**. The **L** fader corresponds to the left channel and the **R** fader to the right channel. Adjust the loop length as multiples of the value set with the corresponding time unit selector located directly beneath the faders.
- (3) **L and R (B)**: Click and drag to adjust the length of the audio section to be looped in looper **B**. The **L** fader corresponds to the left channel and the **R** fader to the right channel. Adjust the loop length as multiples of the value set with the corresponding time unit selector located directly beneath the faders.
- (4) **A/B**: Click to switch between two parallel loops, **A** and **B**. Select one of the two to send it to the output.
- (5) **Time unit (A)**: Click and drag to select the base time value for looper **A**. The set value is multiplied by the **L** and **R** (**A**) controls.
- (6) **Time unit (B)**: Click and drag to select the base time value for looper **B**. The set value is multiplied by the **L** and **R** (**B**) controls.

(7) **Window**: Turn to set the shape of a windowing function that is applied to the loop. Turn it fully right to set the fade-in and fade-out at the loop boundaries to be short and steep, retaining most transients located near the loop start at the expense of possible clicks at the loop boundaries. Turn it fully left to set the fades to be longer and to have an exponential shape, which results in excellent suppression of clicks but may also reduce “punch” and subjective loudness. In-between knob positions interpolate fade lengths, shapes and some other under-the-hood aspects of the “window” function non-linearly. Adjust this control by ear—whatever sounds better IS better!

(8) **Smooth**: Turn to apply a smooth crossfade to the switching between loopers A and B, as well as to the onset of looping when **Freeze** is activated.

(9) **X.gate**: Use this to define the behavior of the effect unit when the **Freeze** button is inactive. Activate **X.gate** to pass the input through to the output while **Freeze** is inactive. Deactivate **X.gate** to mute the output, unless the **Freeze** button is active.

(10) **Gate**: Turn to apply or disengage a gater effect to the looped playback. In the center position, this circuit is off and no gating is applied. Turn it to the right to mute audio at the end of the loop: the higher the value, the earlier inside the loop the muting takes place. Set fully right, there will be only a short snippet playing back at each loop start. Turn the control left of center position to apply the loop length of looper A to looper B and vice versa. This way the gating pattern is of different length than the looping pattern, which creates interesting poly-rhythmic structures.

9.4.6 Iteratron

A repetition, filtering and amplitude-modulation based looper effect. A segment of the input signal is sampled when the **Freeze** button is clicked and held (or its assigned modulator passes a threshold) and played back as a loop until the button is released (or the modulator drops below the threshold). During the looped playback, the signal may be amplitude modulated in stepped rising or falling shape, where the number of steps is set using the **Steps** control and their duration by the **Repeat** control (so each step is exactly as long as one loop cycle). This stepped pattern will repeat for as long as the **Freeze** button is active. Similarly, you can apply high-pass-, bandpass-, or low-pass filters, and apply a stepped modulation pattern with a number of steps set by **Steps** and step-length set with **Repeat** etc. You can also invert the amplitude or filter modulation patterns thus created for the right channel.



Iteratron

(1) **Freeze**: Activate to apply the looping. When off, the output is either silent (when [X.gate](#) is deactivated), or the input signal is passed through unaltered (when [X.gate](#) is activated).

(2) **Steps**: Turn to select how many steps (of the length set with [Repeat](#)) the Filter and Amp Shape patterns have.

(3) **Repeat**: Turn to select the length of audio that is captured when Gate goes high.

(4) **Amp shape**: Slide to adjust the depth of the applied amplitude "shape" pattern set with the [Steps](#) and [Repeat](#) controls. Slide [Amp](#) shape fully left to decrease the level logarithmically from unity gain down to zero (fade-out) over the repetition length set with [Steps](#) and [Repeat](#). At center position, no amplitude modulation is applied. Slide it fully right to increase the level logarithmically from zero to unity gain (fade-in) over the repetition length set with [Steps](#) and [Repeat](#).

(5) **Amp Pan Invert**: Activate to apply left/right inversion for the Amp Shape. When this is activated and the Amp Shape slider is set to fade-out (left), the left channel will be fading out while the right channel will be fading in.

(6) **Filter**: Click to switch the filter on or off.

(7) **Filter shape:** Slide this to adjust the depth of the applied filter Shape pattern. Slide it fully left to set the cutoff frequency of the filter to an initially high frequency that decreases over the repetition length set with [Steps](#) and [Repeat](#). In the center position, no cutoff modulation is applied. Slide it fully right to set the cutoff frequency to start at a low frequency that increases over the repetition length set with [Steps](#) and [Repeat](#).

(8) **Filter Pan Invert:** Activate to apply left/right inversion for the [Filter](#) shape.

(9) **Filter type:** Select one of three filter types:

- [HP](#): Click to set the topology of the filter to high pass.
- [BP](#): Click to set the topology of the filter to bandpass.
- [LP](#): Click to set the topology of the filter to low pass.

(10) [Freq](#): Turn to adjust the base cutoff frequency of the filter.

(11) [Reso](#): Turn to adjust the amount of resonance for the filter.

(12) [Smooth](#): Turn to apply smoothing to the Filter and Amp Shape patterns' steps.

(13) [X.gate](#): Use this to define the behavior of the effect unit when the [Freeze](#) button is inactive. Activate [X.gate](#) to pass the input through to the output while [Freeze](#) is inactive. Deactivate [X.gate](#) to mute the output, unless the [Freeze](#) button is active.

9.4.7 Reverseoid

A reversing effect that repeatedly captures a segment of audio and then plays it back in reverse after capture. The parameters of this effect unit are not available for modulation, with the exception of those found in the footer.



Reverseoid

- (1) **Left**: Turn to set the capture/reversing duration for the left channel.
- (2) **Left dotted note**: Activate to multiply the **Left** value by 1.5, resulting in a dotted version of the note value set with **Left**.
- (3) **Right**: Turn to set the capture/reversing duration for the right channel.
- (4) **Right dotted note**: Activate to multiply the **Right** value by 1.5, resulting in a dotted version of the note value set with **Right**.

9.4.8 Ryuchi

A lo-fi pitch shifter that features independent simultaneous upwards and downwards shifting of unusual topology as well as song position synchronization of the shift buffers. The downward shift is realized by decreasing the increment value of the playback sample counter to non-integer, smaller-than-one values, resulting in a crude downsampling. The counter is reset at rhythmic intervals synchronized to the song position. The upward shift is realized by lowering the loop length of the playback sample counter. A dual windowing approach is used for click-less operation, and reset rate and fade-in/-out rates can be set. Think of this as a circuit-bent pitch shifter that allows up and down shifting simultaneously, while always staying in sync.



Ryuchi

(1) **Step Size**: Turn to adjust the size of the block of audio used; this affects windowing behavior as well as synchronization. Be sure to check out all the different values to hear the differences!

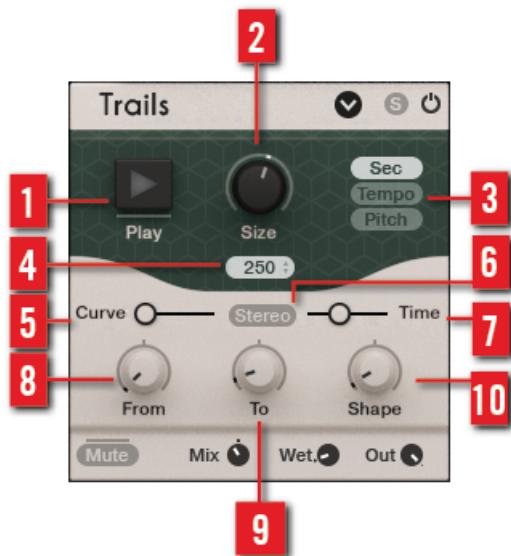
(2) **Fade**: Turn to adjust the amount of windowing (fade-in & fade-out) applied to the audio blocks. Use this to remove any clicks or similar artifacts.

(3) **Up Shift**: Turn to adjust the amount of upwards pitch shifting. Raising this control shortens the length of a looped playback buffer to achieve the effect.

(4) **Down Shift**: Turn to adjust the amount of downward pitch shifting. Test different values to achieve a lo-fi digital sound.

9.4.9 Trails

Whenever a trigger is received, the input signal is sampled for a duration that can be set in seconds, rhythmic values or note pitch, and the sampled signal is played back looped for an adjustable amount of time. For this duration, an AD envelope is calculated. You can adjust its attack and decay times and apply the envelope to the amplitude. Additionally, a window function that has the length of the sampled audio is applied. You can modulate its shape to create effects similar to pulse-width modulation. The shape can also be modulated by the AD envelope, and AD modulation of shape can be inverted for one of the two channels. Eight voices are provided, and triggered cyclically, so the generated “trails” may overlap.



Trails

(1) **Play**: Click and hold (or trigger via modulation) to sample and loop a chunk of audio. The length of the chunk is defined by the **Time** parameter. If you trigger this while a loop still is playing, the next voice will be triggered. There are eight voices available; once the eighth voice is playing, the next trigger will reset the first voice etc.

(2) **Size**: Turn to adjust the length of the audio chunk, in units of seconds, multiples of a note value or pitch (the length of one wave cycle at the set pitch), depending on the state of the **Sec/Tempo/Pitch** selector.

(3) **Sec/Tempo/Pitch**: Use this to select the unit of time for the **Size** control. **Sec** is in seconds, **Tempo** in note values, and **Pitch** in note number (actually the length of one wave cycle at that pitch).

(4) **Note value**: Click and drag to select the base note value for adjusting **Time** when in **Tempo** mode.

(5) **Curve**: Slide to adjust the time that the input audio chunk is looped for, as well as the duration of the calculated AD envelope.

(6) **Stereo**: Click to switch on and apply the AD envelope inversely to the right channel **Shape** setting, creating nice stereo effects.

(7) **Time**: Slide to adjust the shape of the AD envelope. Slide this fully left to set the attack segment to nearly zero and to set the decay to fill duration of the envelope. At center position, the attack segment is 50% of the length of the envelope, and the decay segment to 50% as well. Slide this fully right to set the attack to nearly 100% of the duration of the envelope and the decay to nearly 0 %.

(8) **From**: Use this together with the **To** control to adjust the AD envelope to additionally modulate the **Shape** parameter. The AD envelope is added as an offset to the **Shape** control. The **From** control sets the offset value that is added when the AD envelope has its minimum value, and the **To** knob sets the offset that is added when the AD is at its maximum value.

(9) **To**: Use this together with the **From** control to adjust the AD envelope to additionally modulate the **Shape** parameter. The AD envelope is added as an offset to the **Shape** control. The **To** control sets the offset value that is added when the AD envelope has its maximum value, and the **From** knob sets the offset that is added when the AD is at its minimum value.

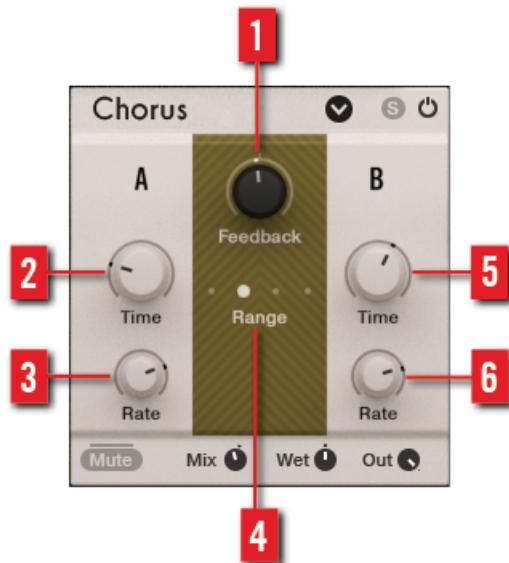
(10) **Shape**: Turn to adjust the windowing applied to the looped chunk. For short chunks, modulating this parameter creates a sound very similar to pulse-width modulation.

9.5 DSP 3 Effect Units

The DSP located third from left contains some more traditional-leaning modulation effects, e.g. chorus and phaser/flanger, but also some creative effects that can be used to alter your sound in more unexpected ways.

9.5.1 Chorus

A chorus-type effect using a stereo 3-tap delay and two triple-sine oscillators with 120 degree phase offset for the sine waves.



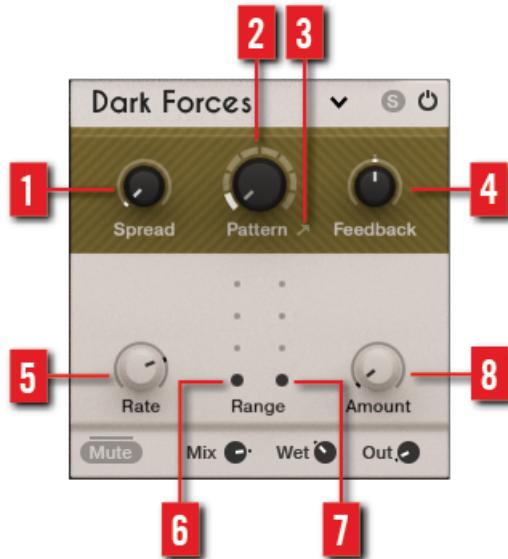
Chorus

- (1) **Feedback**: Turn to adjust the amount of feedback. Turning left introduces phase-inverted feedback.
- (2) **Time (A)**: Turn to adjust the delay time for chorus circuit A.
- (3) **Rate (A)**: Turn to adjust the rate for the modulation LFO.
- (4) **Range**: Click one of four states to select the range for the delay time.
- (5) **Time (B)**: Turn to adjust the delay time for chorus circuit B.
- (6) **Rate (B)**: Turn to adjust the rate for the modulation LFO.

9.5.2 Dark Forces

An effect that has three different modes of operation. All three modes are based on modulated delay lines, akin to a pitch-shifter topology, but deliberately “broken” for unusual effects. **Demons** mode features four quadrature oscillators running at integer multiple frequencies, modulating the delay times of one 4-tap delay each. **Witch** mode features a quadrature ramp oscillator modulating the delay times of four delay taps, and the taps are sequentially crossfaded be-

tween. [Zombie](#) mode is similar to Witch, but the taps are mixed rather than faded, and the oscillator uses sine/cosine waves. All three modes allow the delay times to be set using a spread parameter and also allow scaling the amplitude and adjusting the pan balance of the individual taps using a pattern editor. Also worth noting is that the modulation oscillators go well into the audio range.



Dark Forces

- (1) [Spread](#): Turn to adjust the spacing of the taps, i.e. the delay between taps.
- (2) [Pattern](#): Turn to select one of eight patterns. Each pattern contains level and panning settings for all four taps. Click the arrow icon to open the pattern editor.
- (3) **Arrow icon**: Activate the arrow to open the pattern editor at the center of the interface. For detailed information about the pattern editor, see section below.
- (4) [Feedback](#): Turn to adjust the amount of feedback. Turning left introduces phase-inverted feedback.
- (5) [Rate](#): Turn to adjust the modulation rate. The value is multiplied by the state of Rate Range.

(6) **Rate Range**: Click one of four states to select the range for the **Rate** control.

(7) **Amount Range**: Click one of four states to select the range for the **Amount** control.

(8) **Amount**: Turn to adjust the modulation amount. The value is multiplied by the state of **Amount Range**.

Dark Forces Pattern Editor



Dark Forces pattern editor

(1) **Levels**: Click and drag the sliders to adjust the levels for each of the four taps individually.

(2) **Pans**: Click and drag the bipolar sliders in either direction to adjust the panning for each of the four taps individually.

(3) **Demon**, **Witch**, and **Zombie**: Select one of the three algorithms. For details, see section [↑9.5.2, Dark Forces](#) above.

(4) **Patterns**: Click either button to access the values of the corresponding pattern.

(5) **Copy**: Click to copy the displayed values.

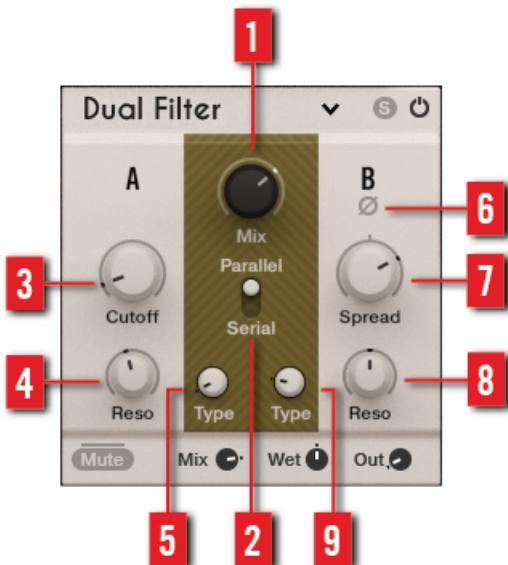
(6) [Paste](#): Click to paste values placed in the clipboard.

(7) [Random](#): Click to randomize all values.

(8) [Close](#): Click to close the pattern editor.

9.5.3 Dual Filter

A dual multi-mode filter (A and B) that lets you choose serial or parallel configuration.



Dual Filter

(1) [Mix](#): When the routing switch is set to [Parallel](#), turn [Mix](#) to crossfade between [A](#) and [B](#); when in [Serial](#) mode, turn to crossfade between [A](#) and [A](#) through [B](#).

(2) [Parallel/Serial](#): Switch to select either parallel or serial routing for the two filters.

(3) [Cutoff](#): Turn to set the cutoff frequency for filter [A](#).

(4) [Reso \(A\)](#): Turn to set the resonance amount and steepness for filter [A](#). Turn fully left to set steepness to 12db/oct and fully right for 24db/oct.

(5) **Type (A)**: Turn to crossfade from low-pass filter (fully left) via bandpass filter at center position to high-pass filter (fully right) for filter A.

(6) **∅**: Activate to invert the phase for filter B.

(7) **Spread**: Turn to sets the cutoff frequency for filter B as offset to the value set with A's **Cutoff**.

(8) **Reso (B)**: Turn to set the resonance amount and steepness for filter B. Turn fully left to set steepness to 12db/oct and fully right for 24db/oct.

(9) **Type (B)**: Turn to crossfade from low-pass filter (fully left) via bandpass filter at center position to high-pass filter (fully right) for filter B.

9.5.4 Filterbank

A filter bank consisting of five 24db bandpass filters. Filter frequencies are set using a base frequency and a spread value. Independent amplitude control is available for each of the bands.



Filterbank

(1) **Cutoff**: Turn to set the base frequency of the filter bank. The value set with this control is also the cutoff frequency of the first filter.

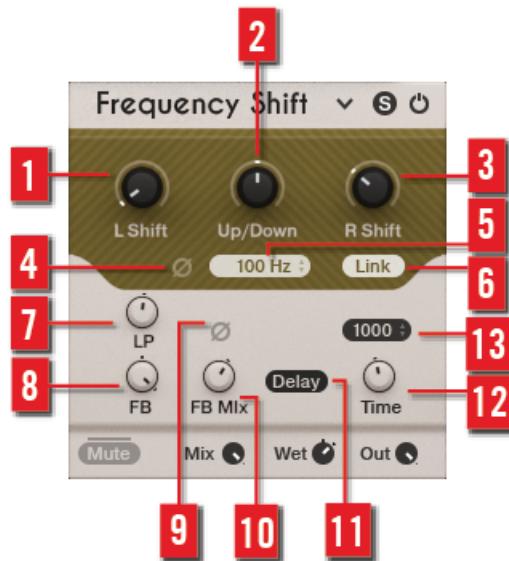
(2) **Reso**: Turn to adjust the resonance amount for all filters simultaneously.

(3) **Spread**: Turn to adjust the spread between the five bands in semitones. For example: when **Spread** is set to 12 semitones, band #2's cutoff frequency will be one octave above the cutoff frequency of band #1, band #3 one octave above band #2 etc.

(4) **Levels**: Slide to adjust the corresponding level of each of the five bands.

9.5.5 Frequency Shift

A stereo frequency shifter with separate up/down shift mix ratios for the output and feedback paths, as well as the option to invert ratios across channels. A modulation delay is available inside the feedback path, and the feedback may be regular or cross-patched.



Frequency Shift

(1) **L Shift**: Turn to adjust the frequency-shifting amount of the left channel.

(2) **Up/Down**: Turn to adjust the up-shifting/down-shifting mix for the frequency shifter's output.

(3) **R Shift**: Turn to adjust the frequency shifting amount of the right channel.

(4) **Ø (Inversion)**: Activate the icon below the **L Shift** control to invert the **Up/Down**-shift mix of the output for the right channel. If **Up/Down** is set to 100% **Up** (fully left), the right channel will be 100% **Down** when this switch is on.

(5) **Shift range**: Click and drag to select the shifting range for the **L Shift** and **R Shift** controls. The available values range from **1 - 10000 Hz**.

(6) **Link**: Activate to override **R Shift** and use the shift-amount value set with the **L Shift** control for both channels.

(7) **LP**: Turn to adjust the cutoff frequency of a low-pass filter in the feedback path.

(8) **FB**: Turn to adjust the feedback amount. Negative values (left of center position) cause a cross-feedback.

(9) **Ø (Feedback inversion)**: Activate the icon above the **FB Mix** control to invert the **Up/Down**-shift mix used for the feedback path for the right channel. If **FB Mix** is set to 100% up (fully left), the right channel will be 100% down when this switch is on.

(10) **FB Mix**: Turn to adjust the up-shifting/down-shifting mix used for the feedback path.

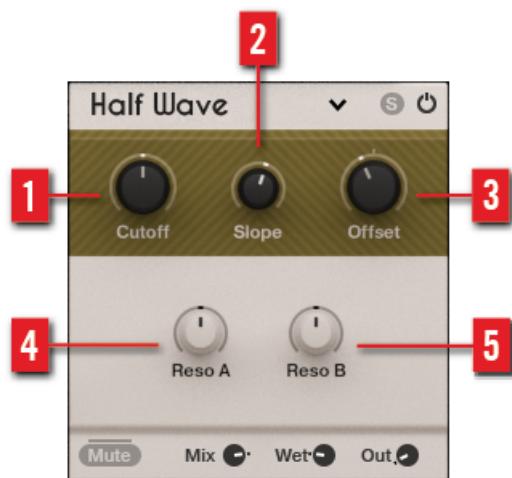
(11) **Delay**: Activate to introduce a delay in the feedback path.

(12) **Time**: Turn to adjust the delay time of the delay in the feedback path.

(13) **Time range**: Click and drag to adjust the delay time range for the **Time** control in milliseconds. Available values range from **25 - 1000 ms**.

9.5.6 Half Wave

A dual bandpass filter (A and B) where the two filters act on positive and negative half-waves of the input signal, respectively.

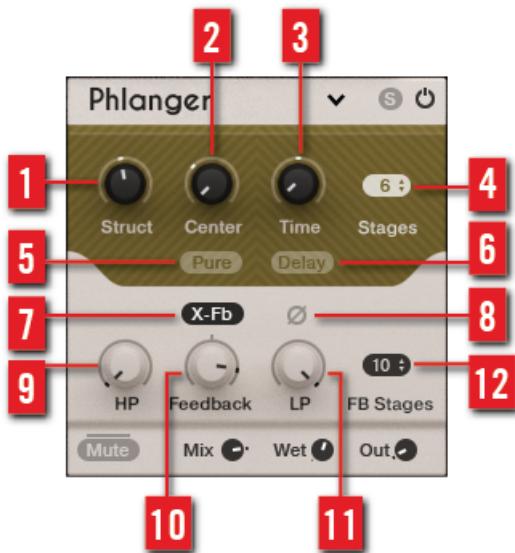


Half Wave

- (1) **Cutoff**: Turn to set the cutoff frequency of filter A.
- (2) **Slope**: Turn to adjust filter steepness for both filters from 12db/octave at fully left to 48db/octave at fully right.
- (3) **Offset**: Turn to adjust the cutoff of filter B as an offset to filter A's **Cutoff** value.
- (4) **Reso A**: Turn to adjust resonance amount for filter A.
- (5) **Reso B**: Turn to adjust resonance amount for filter B.

9.5.7 Phlanger

A phaser with up to ten all-pass stages and a modulation delay in the feedback loop. Essentially it is a hybrid between phaser and flanger. You can select the number of the all-pass stage that is routed to the output as well as which stage is used for the feedback loop.



Phlanger

- (1) **Struct**: Turn to adjust the topology of the all-pass filters.
- (2) **Center**: Turn to set the center frequency for the all-pass filters.
- (3) **Time**: Turn to set the delay time in milliseconds.
- (4) **Stages**: Click and drag to select the number of all-pass filter stages. Available amounts are 1 through to 10.
- (5) **Pure**: Activate to pass the all-pass output directly to the effect's output. When off, it is mixed 50/50 with the input signal.
- (6) **Delay**: Activate to introduce a delay to the feedback path.
- (7) **X-Fb**: Activate to feed the feedback back cross-channel. I.e., the right channel is fed back into the left channel and vice versa.
- (8) **∅**: Activate to flip the phase of the all-pass filter output before it is mixed with the input signal.
- (9) **HP**: Turn to adjust the cutoff frequency of a high-pass filter in the feedback path.

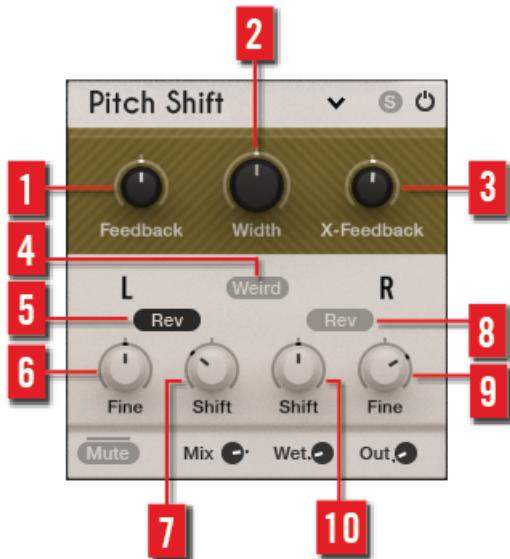
(10) **Feedback**: Turn to adjust the amount of feedback. Turning left introduces phase-inverted feedback.

(11) **LP**: Turn to adjust the cutoff frequency of a low-pass filter in the feedback path.

(12) **FB Stages**: Click and drag to select which all-pass filter stage that feeds the feedback path.

9.5.8 Pitch Shift

A dual pitch shifter with independent pitch and reverse controls for the two shifting units, as well as a “weird” mode that cross-patches between the two shifters.



Pitch Shift

(1) **Feedback**: Turn to adjust the feedback amount.

(2) **Width**: Turn to adjust stereo width (in a “weird” and interesting way!).

(3) **X-Feedback**: Turn to adjust the cross-feedback amount (left into right; right into left).

(4) **Weird**: Activate to create “weird” cross-patching between the two shifters. Think circuit bending!

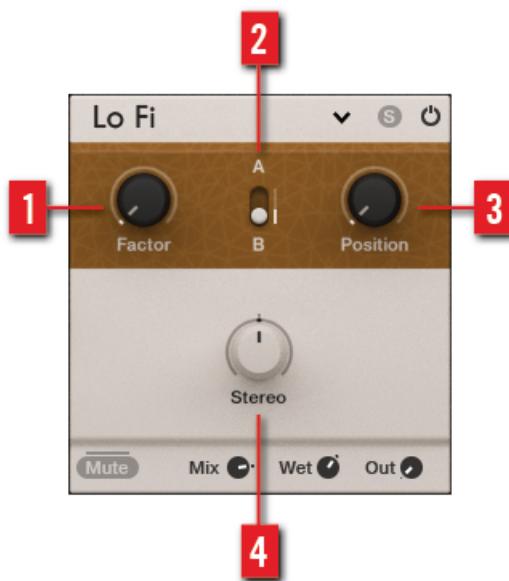
- (5) **Rev (L)**: Activate to reverse buffer playback for the left shifter.
- (6) **Fine (L)**: Turn to adjust the shifting amount for the left shifter in fine increments.
- (7) **Shift (L)**: Turn to adjust the shifting amount for the left pitch shifters in coarse increments.
- (8) **Rev (R)**: Activate to reverse buffer playback for the right shifter.
- (9) **Shift (R)**: Turn to adjust the shifting amount for the right pitch shifter in coarse increments.
- (10) **Fine (R)**: Turn to adjust the shifting amount for the right shifter in fine increments.

9.6 DSP 4 Effect Units

The right-most DSP contains effect units that you can use to distort and add interesting textures to the signal.

9.6.1 Lo Fi

A sample rate reduction algorithm with two unusual modes. Mode A uses four sample-and-hold circuits clocked at octave intervals, which are sequentially subtracted from each other. In mode B, there are also four sample-and-hold circuits clocked at octave intervals, but the subtraction order is inverted. In both modes, sequential crossfading between the four streams is applied and adjustable.



Lo Fi

- (1) **Factor**: Turn to adjust the base downsampling factor.
- (2) **A/B**: Switch to select between mode **A** and **B**.
- (3) **Position**: Turn to crossfade through the four generated resampling streams sequentially.
- (4) **Stereo**: Turn to adjust stereo widening.

9.6.2 Modulo Fry

A three-stage distortion effect consisting of a modulo-based distortion, a rectification circuit with self-modulation options and a hard-clip at the output. A prefilter and a crossfade between various points in the signal path are available.



Modulo Fry

- (1) **Pre LP**: Turn to adjust the cutoff frequency of a low-pass filter at the input of the effect.
- (2) **Modulo**: Turn to set the value that the modulo distortion circuit uses for "destroying" your audio.
- (3) **Reso**: Turn to adjust the resonance amount of the low-pass filter.
- (4) **Self Mod**: Turn to increasingly modulate the rectified signal with the input signal, resulting in very unpredictable but cool sounds.
- (5) **Fry Gain**: Turn to set the gain going in to the "fry" circuit (essentially, a rectification-based distortion).
- (6) **Fry Mix**: Turn to crossfade between the Modulo output and the Fry output. Note that the output gain then feeds into a hard clipper as an additional distortion stage!

9.6.3 Slam Dunk

A compressor—or transient shaper—with negative ratios and unique circuitry specialized for compressing the living daylights out of stuff. Generally, the compressor separates transient amplitude segments from sustained material based on the derivate of the RMS curve. It then uses the transient amplitude curve to either reduce peaks (positive ratios) or attenuate sustain (negative ratios).



Slam Dunk

- (1) **Input:** Turn to adjust the level of the input level.
- (2) **Ratio:** Turn to crossfade from transient curve (fully left) via unprocessed at center position to compressed curve (fully right).
- (3) **Make Up:** Turn to adjust make-up gain. In the signal flow, this takes place before the saturation circuit and thus can be used to drive the signal into deeper saturation when **Sat** is activated.

(4) **HP**: Activate to apply high-pass filtering to the compressor side-chain, so that low frequencies are ignored in level computation.

(5) **Sat**: Activate to add a 4x oversampled soft-clipping saturation stage to the output, creating warmer tones.

(6) **Thresh**: Turn to set the compressor threshold.

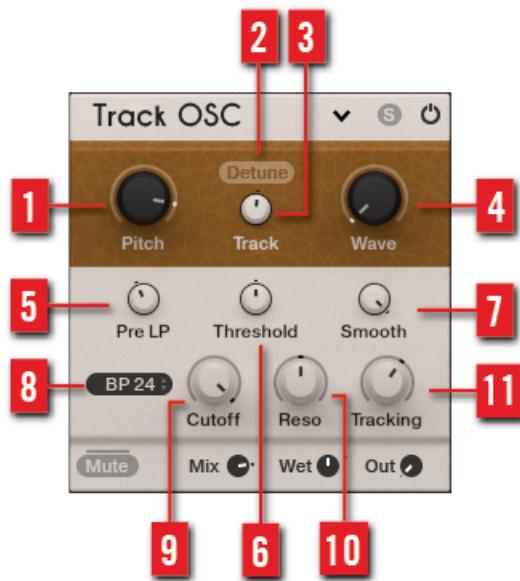
(7) **GR Limit**: Turn to set a limit to the amount of applied gain reduction. Turn fully left for no gain reduction to be possible. Turn fully right for infinite max gain reduction (signal will be nearly off). With **Ratio** set to transient mode, **GR Limit** sets the peak amplitude for the signal.

(8) **Attack**: Turn to adjust the time and curve shape of the attack.

(9) **Release**: Turn to adjust the time and curve shape of the release.

9.6.4 Track OSC

The DSP's input signal's pitch is tracked and used to control a variable waveform oscillator. The oscillator then passes through a filter that may also track the input pitch, and which is fed into a saturation circuit.



Track OSC

- (1) **Pitch**: Turn to adjust the base pitch for the oscillator.
- (2) **Detune**: Activate to apply a slight detuning between the left and right channels for a subtle "fatness."
- (3) **Track**: Turn to adjust how much the input pitch is tracked. At zero (fully left), pitch stays static and doesn't follow the input pitch. At a value of one (center), the input pitch is fully followed. At a value of two (fully right), the input pitch intervals are doubled and followed.
- (4) **Wave**: Turn to adjust the waveform of the oscillator. Set it fully left to produce a sawtooth. At 50% (center position), a square wave is produced. From there upwards, the pulse width of the waveform is reduced until it is nearly zero when the control is set to fully right.
- (5) **Pre LP**: Turn to adjust the cutoff frequency of a low-pass filter in the pitch-tracking section. Set this low if you're hearing erroneous octave jumps.
- (6) **Threshold**: Turn to control the threshold of a gate in the analysis section and the audio path.

(7) **Smooth:** Turn to apply smoothing to the pitch-tracking control signal before it is used to modulate the oscillator and tracking filter.

(8) **Filter mode:** Click and drag to select the type of the filter.

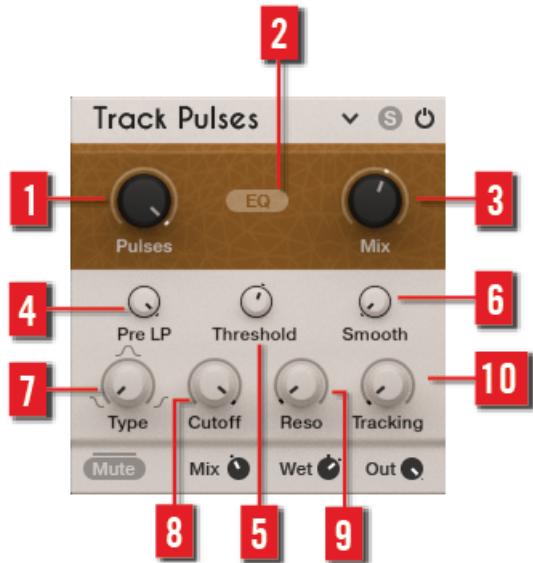
(9) **Cutoff:** Turn to adjust the base frequency of the filter

(10) **Reso:** Turn to adjust the resonance amount of the filter.

(11) **Tracking:** Turn to adjust the behavior of the filter cutoff frequency when following the detected pitch. Set it fully left to not follow the pitch. Set it fully right to let the cutoff frequency follow the pitch fully.

9.6.5 Track Pulses

Whenever the input signal contains a positive-going zero crossing, a one-sample pulse is generated. Four suboctaves are created using flip-flop division, and it is sequentially crossfaded between the octaves to generate the “pulse” signal. This is then cross-mixed with the input signal and fed through a filter that can be set to track the pitch of the first-generation pulse.



Track Pulses

- (1) **Pulses**: Turn to sequentially crossfade through four octave transpositions of pulses.
- (2) **EQ**: Activate to apply a static EQ that makes the pulse signal sound more filled-out.
- (3) **Mix**: Turn to crossfade between the pulse signal and the input signal. Set it fully left to hear only the pulse signal; set it fully right to hear only the input signal. The mix of the two is sent to the tracking filter.
- (4) **Pre LP**: Turn to adjust the cutoff frequency of a low-pass filter in the pitch-tracking section. Set this low if you're hearing erroneous octave jumps.
- (5) **Threshold**: Turn to control the threshold of a gate in the analysis section and the audio path.
- (6) **Smooth**: Turn to apply smoothing to the pitch-tracking control signal before it is used to modulate the tracking filter.
- (7) **Type**: Turn to adjust the filter type, crossfading from low-pass (fully left) through bandpass (center-position) to high-pass (fully right).
- (8) **Cutoff**: Turn to adjust the base frequency of the filter
- (9) **Reso**: Turn to adjust the resonance amount of the filter.
- (10) **Tracking**: Turn to adjust how much the filter cutoff frequency follows the detected pitch. Set it fully left to not follow the pitch. Set it fully right to let the cutoff frequency follow the pitch fully.

9.6.6 Wave Fold

A 16-stage wave-folder that allows adjusting the balance between the positive and negative signal components.



Wave Fold

Fold: Turn to adjust the wave-folding amount.

Balance: Turn to adjust the relative level of the positive and negative parts of the waveform.

10 Credits

Instrument and Product Concept: Denis Goekdag

Product Design: André Estermann

Realization: Dietrich Pank, Alexander Potekhin aka lazyfish

Development: Clement Destephen, Francisco Garcia

GUI Design: Efflam Le Bivic

Quality Assurance: Frank Ellendt, Frank Stempka

Sound Design: Jamil Samad, Denis Goekdag, Richard Devine, Robert Linke

Documentation: Gustav Sto. Tomas

Project Management: Marin Vrbica

©2014 Native Instruments